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Correction

Les pages du fascicule 4 ont été numérotées par mégarde de 403 à 450 au lieu de 471 à 518. On est prié de rectifier la pagination de ce fascicule.

Correction

By an oversight, the pages of Part 4 were numbered 403—450 instead of 471—518. Please correct these page numbers.

Berichtigung

Versehentlich erhielt das Heft 4 statt der Seitenzahlen 471—518 die Seitenzahlen 403—450. Wir bitten, die Seitenzahlen dieses Heftes zu berichtigen.

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REPRINT

FOR

ACUSTICA

OF THE

PROCEEDINGS OF THE FIRST ICA-CONGRESS

ELECTRO-ACOUSTICS

EDITED BY

C. W. KOSTEN

AND

M. L. KASTELEYN

1953

NETHERLANDS

ACUSTICA

Vol. 4 1954 No. 1

INTRODUCTION

This issue, the first number of volume 4, has a special character in that it is a reprint of the "Proceedings of the FIRST ICA-CONGRESS", held in the Netherlands from 16th ··· 24th June 1953. The original "Proceedings" were distributed only among the Congress members. The Congress Committee as well as the Editors of Acustica agreed unanimously that a reprint in Acustica is worth while, thus giving the "Proceedings" a wider circulation and maintaining them as current literature.

Since proceedings are not a compilation of a great many incoherent contributions, the character of this issue deviates somewhat from the usual one. The contributions are grouped together in sections, each covering the activities of the relative "Sections" of the Congress. In order to keep the extent of the "Proceedings" within acceptable limits the text of the contributions had to be very concise. On the one hand this may sometimes cause difficulties for the reader, on the other hand a very lively survey is given which enables the reader to get a good idea of the recent developments in the field of electro-acoustics.

In order to enable a rapid orientation it was thought desirable not to omit the original contents and authors' index. They will not be inserted again in the indices of the entire Acustica volume 1954.

At the very end of the "Proceedings" a membership list can be found which may be of use to many readers of Acustica as it contains full addresses of the authors.

The Editor in Chief

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FOREWORD

THESE "PROCEEDINGS" COVER THE FIRST ICA-CONGRESS held in the Netherlands from 16th to 24th June 1953. The Congress was organised by the Netherlands Acoustical Society "Geluidstichting", which appointed a Congress Committee ad hoc.

The subject matter, electro-acoustics, was divided into six sections (see page 8). Six leading acousticians were invited to preside over a section and to give a "General Review" of the field covered by the section. These General Reviews were read in combined sessions for all members of the Congress. An *international symposium on sound insulation of lightweight structures* was held during the Congress period. For convenience it had been inserted in the programme as a special section, also under the presidency of a General Reviewer. As many as 83 short communications were read in concurrent sessions of the sections, with the General Reviewers in the Chair. The sessions took place in two adjacent laboratories of the Technological University of Delft, viz. the Laboratory for Technical Physics, where the Congress Centre had been arranged, and the Laboratory for Electrotechnics. The texts of the General Reviews, the short communications and most of the discussions have been collected in this book.

In order to maintain the valuable contents of the "Proceedings" as current literature they will be reprinted in Acustica as the first issue of 1954.

It seems appropriate to put some of the other activities on record too. The Opening Session (16th June), held in the famous Knights' Hall in The Hague, consisted of the presidential address of welcome, the formal opening of the Congress by His Excellency the Minister of Education, Arts and Sciences and the opening lecture by Prof. R. H. BOLT (see page 11). After the Opening Session members and their associates were received by the Government and the Municipality of The Hague.

The Municipality of Delft received the Congress members and their associates in the evening of 17th June. The Congress Dinner and Dance was held at Scheveningen on 18th June and was attended by most members.

Two whole-day excursions formed part of the programme, viz. on 19th June to the "Nederlandse Radio-Unie" at Hilversum and Philips Phonographic Industries at Baarn and on 22nd June to Philips Industries at Eindhoven. A short report of these excursions is to be found on page 297.

The Closing Session was held at Delft in the University's ancient Aula. Prof. H. B. G. CASIMIR, one of the directors of Philips Research Laboratories, gave the closing lecture (page 295).

The attendance at the Congress was truly international, with 314 members from 22 countries and 65 associate members (see Membership List on page 303).

The Congress Committee were very much gratified by the invaluable help received from all sides (see list of participating bodies, page 6).

Whereas the major part of the financial aid came of course from Dutch sources, it should be gratefully acknowledged here that a considerable support, including a subsidy for the printing costs of these "Proceedings", was granted by UNESCO — the United Nations Educational, Scientific and Cultural Organisation — through the intermediary of IUPAP-ICA.

Out of the long list of those who assisted individually for the benefit of the Congress I mention Miss C. VERSCHOOR who, beside her activity in the planning and management of the Congress Centre, ably organised the Ladies' Programme, Ir. P. A. DE LANGE, organising secretary, and Ir. M. L. KASTELEYN, co-editor of the "Proceedings".

It was thought very desirable to publish the communications as soon after the Congress as possible. This would never have been achieved, but for the linguistic help of Dr. F. CANAC, Marseilles, Prof. dr. E. MEYER, Göttingen, Dr. E. G. RICHARDSON, Newcastle upon Tyne, the French, German and English editors of Acustica respectively, the very valuable assistance by Ing. F. RÜHMANN, Waiblingen, and last but not least the understanding co-operation of the authors themselves.

The President of the Congress

Delft, February 1954. C. W. KOSTEN

ORGANISATION

The *Netherlands Acoustical Society "Geluidstichting"*

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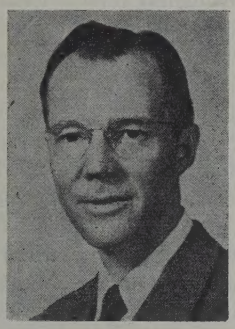
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OPENING LECTURE

by R. H. Bolt
President of the International Commission on Acoustics



We are gathered here on this the sixteenth day of June in nineteen hundred and fifty-three to inaugurate a formal international collaboration in the field of acoustics, and to participate in the first organized activity of that collaboration, an International Congress on Electro-Acoustics. On this occasion, as in any inaugural event, it seems appropriate to review the past and to predict the future. Before deciding to do so, however, one must recognize certain pitfalls. Historical reviews run the danger of becoming tedious, especially for an audience that already knows this history better perhaps than does the speaker. Forecasting the future is equally hazardous, particularly when the audience is the very group whose actions of tomorrow will mould the future that is predicted today.

A particular solution to this dilemma can be found for the subject of acoustics. The evolution of this field has followed a basic thread of significance so impelling that the progression of events from past to present is seen to be an inescapable pattern of progress. Therefore we need not recite in detail the historical developments of acoustics. We need only to cite certain events as landmarks in the unfolding pattern. We need not venture the blueprints of achievements yet to come. We need only to trace the thread through the contributions of today, and these will point unmistakably to the character and role of the New Acoustics of tomorrow.

The fundamental line of thought to which my remarks will be directed is indeed basic, for it deals with inherent characteristics of man and of the physical world in which he lives. Man can hear and man can make sounds with his own vocal mechanism. The physical environment is composed of matter, in solid, liquid and gaseous forms. All matter is a distribution of mass, systematized by forces which in the gross scale we call elastic forces. Because of mass and stiffness, in many combinations, all forms of matter can vibrate and can transmit vibratory mechanical energy or sound waves. But these are the very waves that interact with man, when he listens and when he talks. These aspects together constitute the subject matter of acoustics: vibratory mechanical energy and its interaction with man. In this sense the subject of acoustics is as old as man himself.

The art of acoustics, as distinct from the phenomena which it studies and manipulates, had its beginnings about 2500 years ago, with the work of Pythagoras on the vibrations of strings and the musical effect of the sounds they produce. Under this definition, the study of acoustics is about as old as any of man's intellectual endeavours. Much more important than historical precedent, however, is the manner in which Pythagoras studied his problem. He did not separate physics from engineering. Those, of course, were not his words, but he must have recognized that the question: "How does it work?" was different from the question: "How do you build a better musical instrument?". He did not isolate the subjective from the objective, but rather accepted these as two natural parts of a single problem. In other words, he looked at the problem as a whole.

Let us keep this point of view in mind as we proceed to trace the several lines of development that are converging today to form the NEW ACOUSTICS. This is an integrated field of technology that borrows from and contributes to mathematics and physics, psychology and physiology, electrical engineering and chemistry, medicine, architecture and music. From this melting pot of acoustics there emerge essentially two kinds of alloys. One is a product that serves man directly in his acoustic interactions with the physical environment. The other is a tool or a technique that is useful for research or industrial applications in general. We may think of these latter contributions as by-products that repay in some measure the debt of the NEW ACOUSTICS to the many fields from which it sprung.

Let us introduce one further generalization to which we shall speak. Just what is electro-acoustics, and why is it an appropriate subject for this inaugural International Congress? In the purely technical sense electro-acoustics is that branch of the field that deals with the transduction of energy between its acoustical and electrical forms. By utilizing the electrical form we extend enormously our capabilities for generating, transmitting, receiving and measuring acoustical energy. We can at once employ the versatility of electromagnetic waves and electrons. But in a deeper sense, electro-acoustics is today a major link between the ideas of acoustics and the uses of acoustics. Our physical theories are tested in experiments using electro-acoustic instruments. The applications of acoustics are guided by electro-acoustic measurements, and in many cases the product itself contains electronic equipment and transducers. Electro-acoustics is indeed a common currency for the NEW ACOUSTICS, a logical subject for this Congress.

We have mentioned that PYTHAGORAS was perhaps the earliest practitioner in our field, and that he was interested in musical sounds and instruments. Another contribution of his era was the Greek theatre. Here the unfolding art of drama was brought to the people. Here also the social and political news of the day was communicated to all who gathered on the steeply rising steps. The acoustical requirements for these functions were admirably satisfied by the geometry and by the reflecting surfaces around the stage. There was also some acoustic gadgetry, resonant vases and masks with megaphones to strengthen the words of the actors. To some extent, the actors and orators themselves helped solve the acoustics problem by exaggerating their enunciation and intonation. Thus in the earliest practices of acoustics we see the origin of collaborative efforts in which the architect and the builder, the scientifically-minded philosopher and the orator all combined to enhance the use of sound for purposes of communication and entertainment.

Our pilgrimage stops briefly at the door of VITRUVIUS, whose monumental writings on architecture summarized the practices of the Greek and Roman builders. In these books VITRUVIUS set forth the existing knowledge of building acoustics, integrated in a natural way with the general problems of architectural design and construction.

Through many centuries that followed the Roman civilization, acoustics seems to have been dormant, and then came a gradual awakening of activity in the musical arts. There was much experimentation with musical instruments, with percussive devices, strings, and pipes. New tone colours and dynamics, new methods of construction and techniques of performance were tried in profusion. There were some monsters at least as frightening as any electro-acoustic musical instrument that has yet been devised.

> "The ancestor of the tuba, in the brass choir, for instance, was a thing called appropriately enough, the serpent. It was built of wood, covered with leather, was pierced with finger holes, and in general appearance resembled the death agonies of a boa constrictor. The sounds it emitted, indescribably horrible in themselves, were not even in tune; for the finger holes were placed, not where they would produce notes of the correct pitch, but where the player's fingers could reach them. This same habit of favouring the player's comfort over the laws of acoustics made the flute and clarinet of the 18th and early 19th centuries instruments whose intonation would make a modern symphony subscriber squirm." [1]

Throughout these same centuries the material of music itself was undergoing an evolutionary process. Starting with solo voices or two voices in unison and octave, there were gradually introduced additional voices in fifths, thirds, and fourths. And little by little the harmonic structure of music became more intricate, the palate of the composer more rich. The underlying laws of acoustics, the harmonic properties of strings and pipes, the adaptation of man to consonance and dissonance, were unobtrusively but resolutely at work. In the end products—the organ, the orchestra, the music of today— we cannot distinctly separate the contributions of the craftsmen and the composer, the scientist and the virtuoso, the listener and the musicologist. All of these people have together, through centuries of collaboration, produced today's contributions of acoustics to the musical arts.

By the year 1600 the stage was set for the awakening of physical science. No period in history has had a more profound influence on man's intellectual struggle with the world around him. Intimately entwined with the birth of physics itself was the first thread of physical acoustics. GALILEO in Italy, MERSENNE in France and HOOKE in England were making the first quantitative observations on vibration and sound.

New methods of mathematical analysis were becoming available to describe such phenomena as the propagation of continuous waves in a fluid medium, and soon this new mode of thinking was systematized in the monumental writings of ISAAC NEWTON.

In this same period, while the science of sound was emerging as an integral part of physical science in general, there occurred an event which is particularly appropriate to commemorate today. The infant science of sound was first recognized as a distinct personality and was christened by JOSEPH SAUVEUR, who gave it the name *"acoustics"*.

And now one is strongly tempted to recount the impressive story of the development of physical acoustics from 1700 to 1900, but we have agreed that history, as such, is not our message today. Let us then simply recite in recognition the names of BERNOULLI and EULER, FOURIER and LAGRANGE, CHLADNI and KIRCHHOFF, LAPLACE, POISSON and BOLTZMANN, HELMHOLTZ and RAYLEIGH. Hardly any aspect of mathematics and pure physics was left untouched by these great scientists, and yet it is this same group that has given us nearly all of the principles of the theory of sound. To them there were no arbitrary divisions among physical truths. Their creativeness was built on depth of knowledge and breadth of interest. These same attributes of intellectual vision must be carefully fostered today, in the face of increasingly diverse attractions in applied acoustics, if we are to insure the continued productivity of our field.

We have already noted that the subjective aspects of sound, particularly the aesthetic quality of musical sounds, have aroused considerable curiosity for some 2500 years. Serious attempts to relate the subjective experiences to the mechanism of hearing commenced about the time the science of physics was awakening. Progress was less rapid, however, because it was dependent upon a knowledge of anatomy and physiology, and these subjects were slow to advance in those years. The relation between pitch and frequency was recognized by GALILEO, SAUVEUR and MERSENNE. It was not until about 1800, however, that the frequency limits of audibility were studied, by SAVART, SEEBECK and KOENIG. Shortly thereafter the initial hypothesis on the mechanism of audition was proposed by OHM.

The first comprehensive treatise on the sensation of tone was contributed by HELMHOLTZ, who had a remarkable understanding of the life sciences as well as of physics. His work, together with subsequent contributions by RAYLEIGH and several of his contemporaries, marked the beginnings of physiological and psychological acoustics. During the first quarter of the present century the problems of hearing began to attract the attention of many experimental psychologists and physiologists. Thresholds and differential sensitivities for pitch and loudness were measured. Newly available electronic techniques led to such significant discoveries as those of cochlear potentials and microphonics.

Now, within the last few years, we have come to recognize that all of these facets of subjective acoustics are closely interrelated. With this unifying view there has appeared a new name, "bio-acoustics", to encompass all aspects of the study of man's interaction with his physical environment.

The forward progress of civilization is dependent in large measure on the application of new knowledge to beneficent purposes. We see this process at work in the field of acoustics, in the early use of ear trumpets to aid the hard of hearing and in the interpretation of human sounds of coughing, breathing and heartbeats for the diagnosis of ailments. The stethoscope as a specific diagnostic instrument was devised by a French physician, LAENNEC, about 100 years ago. The advent of the telephone and the vacuum tube led to entirely new electro-acoustic techniques for aiding the deaf and diagnosing the sick. The growing importance of these applications is exemplified in the session on hearing aids and audiometers at this Congress and in current research activity in medical ultrasonics.

The phenomenon of sound is not exclusively beneficent. Unwanted sounds can be a nuisance in the home and in the office, and can interfere with speech communication. Intense noise can damage the hearing mechanism and in other ways adversely affect the health of man. It is, therefore, a duty of the field of acoustics to seek methods for the control of noise and to monitor the ever-increasing noisiness of modern industrial living.

Let us now return to our central theme. We have drawn on certain historical references to examine the pattern of progress in acoustics. In so doing, we have introduced the several elements that combine to form the structure of the NEW ACOUSTICS. This field has a nucleus of two parts: physical acoustics and bio-acoustics. The activity of the nucleus is basic research. On the one side is physical research in the mechanisms of sound generation, transmission and reception, and in the behaviour of matter in the presence of vibratory forces. On the other side is psychological and physiological research in the perception of sound, in the transmission of acoustic information through the nervous system, and in the behavioural response of individuals to the sounds around them.

This dual nucleus feeds the applications of acoustics in four general areas: 1) audio communications, 2) musical arts and entertainment, 3) health, comfort and safety, and 4) techniques for science and industry. The entire structure is served by electro-acoustics, which deals with instrumentation for basic research and with devices that are used in nearly every phase of application.

The area of communications includes the continuing improvement of microphones, loudspeakers and associated components; the analysis and synthesis of speech sounds; the design of rooms for speech; and the development of electrical communication systems. The health and comfort of man are served by hearing aids and audiometers; by the control of noise in buildings and in machines; by planning and legislation for the abatement of environmental noise; and by the application of acoustics to medical diagnosis and treatment. To the musical arts acoustics brings the analysis of musical sounds and their subjective aesthetic effects; the synthesis of new musical instruments; the improvement of music recording and reproducing equipment; and the designing of finer music studios and concert halls.

In the realm of new techniques for science and industry the applications are already so numerous that they touch nearly every field of technology. Sound waves are used in the analysis of chemical substances, in the testing of metals and plastics, in navigation and underwater location, in geophysics, in the cleaning of surfaces and the preparation of materials. New uses for sound are developing so rapidly that the photo-electric "magic eye", with all its industrial accomplishments, will soon be matched in versatility and ingenuity by the "magic ear" of sonics.

These four areas of application, in turn, are linked collaboratively with many other fields of science and engineering. These include physics and chemistry, radio, telephone and electrical engineering in general, music, architecture and structural engineering, medicine, public health, psychology, physiology, metallurgy. We should not infer that the field of acoustics is unique in possessing so complicated a structure, for many fields of technology can similarly be described by a basic nucleus, a limited number of primary functions and a multiplicity of connections with other fields. But the very recognition of this organizational character is one of the significant contributions of the present era. And again, acoustics is among the pioneers in understanding and pursuing an integrated approach to a total problem of man and his environment.

This approach is not without its difficulties. From PYTHAGORAS to RAYLEIGH it was possible for a single individual to master a sufficiently wide scope of the existing knowledge, so that he alone could make significant progress on a problem as a whole. Today this possibility is greatly diminished. Individual contributions are no less important, but they will tend to lie along increasingly specialized lines as viewed against the broad problem.

The modern counterpart of the scientific savant is the research team which combines the requisite knowledge of many subjects. Often this team is a network of individuals or small groups in different cities or countries, working simultaneously on different aspects of a general problem. Essential for success under such circumstances is a thoroughly adequate communication system. The needed communications include individual visits and correspondence, seminars and meetings of societies, and scientific publications.

Progress in acoustics during the last 25 years has been aided greatly by organized activities such as those of the Acoustical Society of America and of acoustics groups and societies in several other countries. Our experiences have shown that scientists and laboratories separated by hundreds or thousands of miles can indeed work as a team if adequate organizational assistance is available.

The greatest responsibility of the NEW ACOUSTICS, as it looks to the future, is to give all possible encouragement to those who pursue basic researches on the interaction of sound with matter and with man. We who carry on the long tradition of acoustics must recognize clearly its basic problems, its multiple structure and its potentialities for service. We must emphasize the importance of broad educational objectives for students with interests in acoustics. Through publications, societies and team efforts we must ease the difficulties of cross-collaboration among many specialized fields. This point of view is given international recognition as we meet here today, as we share our knowledge at this Congress, as we initiate this international affiliation.

The International Commission on Acoustics

The manner in which the International Commission on Acoustics will attempt to fill its role can best be reported by commencing with a brief description of its relation to the United Nations. Working under the authority of the General Assembly, the Economic and Social Council deals with cultural, educational, health and related matters. Even before the founding of the United Nations, there were several intergovernmental agencies, such as the ones with which we are acquainted in the fields of telecommunications

and scientific standards. One of the tasks of the Economic and Social Council is to bring these existing agencies into coordination by negotiated agreements. Another task is to form new agencies as required to extend international collaboration and aid. One of these new bodies was the United Nations Educational Scientific and Cultural Organization, or simply UNESCO.

Under UNESCO, in turn, the affairs are managed by more specialized agencies of which one is the International Council of Scientific Unions, or ICSU. From this Council the authority for official international activities and the financial assistance for these activities flow directly into unions within the several established fields of science. We shall name only the International Union of Pure and Applied Physics, IUPAP. This Union, like others at its level, may establish Affiliated Commissions such as the International Commission on Optics, and it may establish Special Commissions such as the International Commission on Acoustics, ICA.

We would seem to be at the end of a long chain of Command: UN, EcSoC, UNESCO, ICSU, IUPAP, ICA. Our situation might have been even more complicated, however, for it is within the power of the several International Unions to establish Joint Commissions. We could, for example, be established jointly between physics and psychology, or medicine or electrical communications. Such inter-field collaboration would indeed be appropriate in view of the multiple structure of the New Acoustics. But if the field has taken centuries to evolve we can allow a few years to work out carefully such international coordination as will best fit our particular needs.

The International Commission on Acoustics itself is a body with seven members, all of whom are in attendance at this Congress. These members, each from a different country, were appointed by the International Union of Pure and Applied Physics when it established the Commission in the summer of 1951. We held a meeting shortly thereafter in London, continued our discussions in the summer of 1952, and are again meeting during this week. Our main actions to date may be stated briefly as follows:

1) We recognize the inter-disciplinary nature of the field of acoustics as discussed in this address. We have, accordingly, informed the International Union of Pure and Applied Physics that we shall explore informal collaboration with unions in other fields, but that we consider our present status as a Special Commission, fostered under physics, to be appropriate and desirable for an initial period of several years.

2) We have established, through our Secretariat, an international clearing house for notices and schedules of activities of the several national groups and societies in acoustics.

3) We have officially sponsored this first Congress and are planning to sponsor future congresses.

4) We are taking steps to encourage international fellowships and exchange visits among acoustics people in different countries.

5) We desire the simplest possible organization, without duplication and unnecessary procedures, that can effectively foster the continuing advancement of acoustics on an international level.

Now, in behalf of the Commission, I wish to extend greetings to all who are attending this Congress. We express our appreciation to the Honorary Committee, and our gratitude to the Netherlands Participating Bodies for their substantial assistance in making this a noteworthy occasion. And finally, we are deeply indebted to the Netherlands Acoustical Society for accepting the organizational responsibility, and especially to the Organizing Committee for their masterful planning of this, our inaugural International Congress.

References

[1] Taylor Deems, Of men and music, Simon and Shuster, New York 1938.

SECTION I
SOUND RECORDING

GENERAL REVIEW

by R. VERMEULEN
Philips Research Laboratories, Eindhoven, Netherlands



Sound recording has different aspects: the more technical one, that can be dealt with by measurements and calculations, design and invention; a more artistic one that we engineers would like to tackle in the same way, though we should not do so, and an economical one, which we cannot treat in an exact way, but which we still have to master somehow, as it is the decisive one.

We feel most at home when we can discuss facts that can be measured, like:

1) frequency range,
2) response, steady state and transient,
3) non-linear distortion,
4) noise,
5) playing time.

These are quite distinct concepts, that can be evaluated separately, but which we find so intimately interlaced in the design of a sound recording system that to improve one we often have to tackle the other. What limits the frequency response of a gramophone record, for instance, is not the inertia of the mechanical system that cuts the groove, but the combination of the noise of the record material and the geometry of the groove (see paper by F. V. HUNT *). We find again and again that what was intended as an improvement in one respect can be used just as well or even better in removing a deficiency in quite a different respect.

Why were long-playing records for instance made available only after the war and not before? Not so much because of any major invention in the field of disc recording but purely because the introduction of the high-frequency bias gave magnetic recording such an impetus that to meet its competition the gramophone record had to be freed from the bounds that fettered it to its mechanical reproducer. Although we find it hard to believe, even now half the gramophones are still mechanical ones. Perhaps the transistor will take care of these abominable things. As soon as it was deemed no longer necessary for the disc to be able to furnish the whole of the acoustic energy itself, it became possible to apply some improvements which already existed. The only thing that had to be done was to establish new standards. Just how difficult it is to find the one and only standard, however, is best shown by the three speeds of $33\frac{1}{3}$, 45 and 78 rev/min and by the variety of frequency characteristics now in use.

As it is not necessary to exchange, handle and fix the permanent stylus, the latter can be of extremely small dimensions and the mass of the other moving parts of the pick-up can be reduced accordingly. The smaller forces thus required allow a sharper tip of the stylus, even when the latter is guided by a softer material such as plastic. We must not forget that the accelerations of the stylus are enormous, several hundred times that of gravitation, comparable to that occurring when a car moving at 100 km/h is stopped within 25 cm.

The forces were further reduced by allowing the stylus to be free to move in the vertical direction. The smaller transverse forces also allow for a proportional reduction of the vertical force which has

*) This and similar references in this General Review refer to papers, read in Section I, the text of which can be found hereafter.

to keep the tip of the stylus in contact with the record. This force has been reduced in the professional pick-ups to something like 3 grammes weight (= 30 mN).

The sharper needle will fit in a narrower groove, the smaller grooves can now come closer together without intersecting. The groove-pitch can also be reduced and thus we get a longer playing time. The sharper stylus tip, moreover, allows for reproduction of smaller wavelengths; thus the speed of the turntable or the diameter of the inner groove can be reduced without loss of quality.

Wax, formerly the only recording material, requires temperature control and careful handling, and implies the use of heavy discs. It has, however, now been superseded by lacquer. A smoother groove and hence less noise result from electrically heating the cutting-stylus and from the homogeneity of the plastic material into which the grooves are pressed. Negative feedback of the cutter reduces non-linear distortion. Less distortion at high frequencies allows more pre-emphasis. All this results in better quality and longer playing-time. Magnetic recording having become more reliable than disc recording, almost all discs are at present re-recorded from magnetograms which not only can be checked during recording but which can also be corrected and edited far more conveniently. Thus many risks are removed and play-back of the lacquer becomes unnecessary; the lacquer can be softer, and the recording is improved. When re-recording, one can predict the amplitude of the groove and this makes it again possible to lengthen the playing time by adapting the groove pitch automatically to the modulation. F. SCHLEGEL *) will discuss the modern recording in this Section.

We have seen how the development of the gramophone was stimulated by its younger competitive brother, namely the magnetic tape. How is it that the latter did not replace it entirely? It is certainly not inferior in playing-time, frequency range, noise level and in all those things that make for what we have become accustomed to call quality. In other respects, however, there are distinct differences which assign each one to its own task. To produce a gramophone record is professional work; magnetic recording can be done by any amateur. The handling of a tape, however, requires more skill than putting a pickup on a gramophone record, the latter being also more easy to store and to transport. The possibility of erasing a recording and of using a tape more than once is certainly a feature for the amateur but perhaps not so much for the professional. The gramophone record, however, is still unsurpassed in the mass production of cheap copies. To sum up, we can claim that whereas the gramophone is the instrument to reproduce music for the general public, the magnetic tape recorder is above all the instrument for making a recording.

In the motion picture industry the photographic sound-tracks are now also being re-recorded from magnetic tapes. The art of "magnetic striping", i.e. putting magnetic material on photographic film, both raw and processed, is developing rapidly in various ways. This will make sound-on-film accessible to the amateur. The advantages of magnetic over photographic sound-tracks on professsional positive copies will, however, remain difficult to evaluate so long as the large investments in the cinemas all over the world have a deciding influence on the choice. It may be that magnetic tape will be used for the recording of pictures, though some people maintain that light and cathode rays are more suitable and more rational.

It is a well-known fact that the sound record proper does not have a flat *frequency characteristic* and should not have one. This raises two problems: the first one concerns the measurement of the frequency characteristic of the recording proper, the second one the standardization of this frequency characteristic in order to make interchange of records possible.

For the gramophone the first problem is solved by the ingenious method of ERWIN MEYER, which makes the velocity amplitude directly visible. For magnetic tape, the theory of W. K. WESTMIJZE *) is probably sufficiently exact for the evaluation of the recorded flux.

The frequency characteristic must be chosen in such a way that the available signal-to-noise ratio becomes as large as possible. Perhaps I should explain that there really exists a maximum. As long as I can raise part of the recording characteristic, say at the high-frequency end by giving more pre-emphasis, without increasing the distortion, the de-emphasis in the reproduction will lower the noise contained in that frequency region. But as soon as I give so much pre-emphasis that I have to lower the level of the whole signal to prevent distortion, I must increase the amplification in reproduction and the noise in the rest of the spectrum will be raised. Between these two extremes there is an optimum; this depends, however, on the character of the music to be recorded. For a theoretical foundation of the choice we should know not only the probability distribution of the sound-power over the spectrum, but the correlation between the frequency bands as well. These probabilities depend, however, on the weights assigned to the different kinds of music and this explains the existing differences of opinion. It is of great importance, however, that a generally accepted standard be established for gramophone

records as well as magnetograms. In my opinion the best way to do this is to specify a standard reproduction apparatus. It will then be possible to reproduce a recording as it was intended to sound, this being of prime importance to artistic justice.

Non-linear distortion is very annoying in music, not so much on account of the harmonics it generates, but because of the inharmonic combination tones it causes. It is therefore a good and growing practice to measure these combination tones themselves. It is indeed true that we can calculate these combination tones from the harmonics for single-valued functions, but this is impossible when the relation between in- and output is not single-valued and is not independent of frequency. The method proposed by HILLIARD to measure the intermodulation between a very low and a very high frequency is sound, but will not suffice in all cases. To give only one instance: a low frequency in a magnetogram induces only a small voltage and will not cause intermodulation. A high frequency can, however, still overload the first valve. In such cases it is recommended to measure the difference tone of two high frequencies, whereas in other cases we should even explore the whole spectrum.

In magnetic recording a special kind of distortion is encountered; this is perceived as a noise varying with the signal and is called *"modulation noise"*. It can have different origins, to wit: amplitude modulation of the signal by random variation of the sensitivity of the magnetic tape (drop-outs in extreme form) or variation of the flux picked up by the reproducing coil through changes in the distance between tape and head. It is also caused by frequency modulation of the signal when the tape executes rapid longitudinal vibrations. As WERNER has shown, these vibrations are excited in much the same way as the transverse vibrations of a violin string. The oscillations are effectively damped by a small freely rotating roller, which is pressed lightly against the tape. Its inertia prevents it from following the rapid vibrations of the tape (e.g. 5 kc/s) and the friction between tape and roller prevents the vibrations from building up. A better remedy would be to have the friction between tape and head increasing with velocity instead of decreasing as is the case with ordinary dry friction.

It is, however, not quite right to consider modulation noise as a new phenomenon specific to magnetic recording. The ordinary *background noise* of a photographic recording is essentially the modulation noise of the zero-track. This zero-track is indispensable because only positive signals can be recorded. The fact that magnetic tape is able to record negative signals as well as positive ones makes a zero-track superfluous. Thus the modulation noise of the signal is no longer masked by that of the zero-track.

This explains also the low level of the background noise and the noticeable increase produced by any constant magnetization of the tape, caused for instance by a remanent magnetization in a head or in a roller. These must always be carefully demagnetized. The remaining slight background noise is caused (so long as it does not originate in the recording amplifier) by the structure of ferromagnetic materials. These consist always of fully magnetized domains. In the demagnetized state the extraneous fields of these domains primarily cancel each other out but in actual fact do not do so completely. The remaining stray fields, though of microscopic dimensions, induce the inevitable background noise.

The saturated magnetization of these domains has preferred directions and configurations, which minimize the energy and are separated from others by barriers of higher energy. The application of an extraneous field causes a crossing of the barriers, thus leaving the medium permanently magnetized; but not so very permanently, as the barriers can be of very different heights and the energy necessary to cross them can also be supplied by a chance distribution of the heat energy assisted by energy from other extraneous sources. The first consequence is that in the course of time all permanent magnetization will disappear; luckily, however, the time constant can be immense. Barriers of low height will be crossed nevertheless in short times, domains falling into disorder out of their ordered arrangements. Part of the signal is thus converted into noise. On the other hand a very small extraneous field can in the long run cause permanent magnetization. This explains the *"printing"* of the recorded signal from one layer of a reel on to the adjoining ones. Its time-function is logarithmic and as such has the characteristics of being the result of a chance excess of energy i.e. it is enhanced by higher temperature and by small alternating magnetic fields of the order of some oersteds. The lowest barriers are obviously crossed first, it is therefore also possible to erase the print again by a weak magnetic field or by a rise in temperature without disturbing the recording proper which is anchored by higher barriers. The ideal magnetic tape should therefore only have barriers of one and the same height. Such a tape would have moreover a sharply defined biasing field, which makes the domains sweep freely over the barriers.

The above implies that magnetic particles in the tape should be of uniform size as the coercivity depends on the size of particles; the smaller the particles, the higher the coercivity. They should be numerous and be evenly distributed. With particles of 0.1 μm and a layer 15 μm thick, the number that passes a head is of the order of 10^{13} per second,

which is large compared to the number of electrons in the photoelectric current of the photocell of a photographic reproducer. But to have the full benefit of this large number the particles must not agglomerate into large clusters, as they are apt to do on account of their mutual magnetic attraction. We will not enter any further into magnetic recording as the contribution of M. CAMRAS *) will survey the recent development of this art and H. SCHIESSER *) is going to discuss the relevant physical properties of the tape.

In the art of telecommunication there is a new and extremely interesting science which is developing rapidly, namely "*information theory*". Can we expect any help from that quarter? We will have to start with a definition of the information we want to record, and here we meet with insuperable difficulties when we are dealing with music, because the information in music is, as I see it, of quite a different character from the information in speech. When we want to supply only the essential information contained in *speech*, we need not give a true copy of the sound, but we can give it in code. We can record it by writing or printing letter symbols and for rapid communication we can send telegraph or telephone signals. And most important of all, we can measure the efficiency of a system or of a channel numerically although subjectively, by evaluating "articulation" or "intelligibility".

If we knew the character of the essential information contained in a recording of music, then perhaps we could make a numerical judgement of its deficiences. In speech we only want to recognize the meaning of the sentences, in music, however, we are not interested in a record that merely permits us to recognize that the music which is being played is probably the waltz "An der schönen blauen Donau". This sort of information should be printed on a label which is precisely what we always do. The information contained in the score is of greater interest. But I do not think that records are being made as a kind of "book for the blind" for those who have not learned to read a score. If this were the case, it would be senseless to make recordings of the rendering of the same score by different artists. Perhaps the information we look for on a music record may be the deviations from the score: the slight variations in rhythm, intonation and in strength; the rubato's, the accents, agogic and dynamic; the course of a crescendo or diminuendo; all those attributes that make the difference between the rendering by a genius or by any ordinary artist. Perhaps we may say that what we look for in music is what is called "redundancy" in information theory. Perhaps what is annoying in music reproduction is not so much lack of information but the distraction of our attention from the essential information by alien sounds. If this were the case, information theory has nothing to offer us. I am sorry to have to leave this question open. I dare not tread further on this aetherial ground for I fear that this constitutes a problem which is not to be solved by science but by art.

However, as scientific people we cannot do without numerical judgements; the only competent measuring device being the human ear, we are obliged to revert to subjective tests, which will be discussed by T. SOMERVILLE *). In our laboratory we have asked a rather large number of people, most of them interested in music, to give us their opinion as to which one of two reproductions of the same short piece of music sounded the more natural to them. They listened in an arbitrary sequence, to the musicians themselves playing behind a screen and to the reproduction of a magnetic recording, the one following immediately after the other. At the same time we tried to find out if they could distinguish in an ABX test the two renderings. The results are shown in Fig. 1. We can say therefore that there is approximately an even chance that somebody

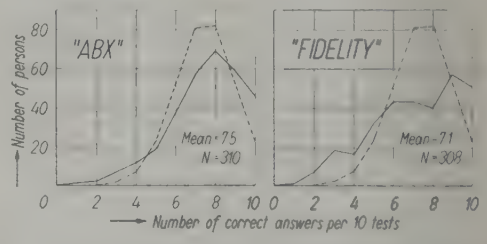


Fig. 1. The lines drawn relate the number of persons to the number of correct answers they gave in a series of 10 tests. In the "*ABX*" test they had to identify the third (*X*) of three performances *A*, *B*, *X* as being the same as either the first one (*A*) or the second one (*B*). In the "fidelity" test the answer was considered correct when they thought the live performance the more natural one. Some 50 persons (16%) were always right. A smaller number were right half the time, i.e. they could not hear any difference and simply guessed. The dotted lines are binomial probability distributions having the same mean. They show that the set of test persons was not homogeneous and that some persons had the tendency to consider in the fidelity test the reproduction more natural than the live performance.
N = total number of persons.

cannot hear the difference between the original and the reproduced music. I am not quite sure, but I have the impression that the frequency characteristic is perhaps not so all important for good music reproduction, as is generally believed. We know that it is possible for a distinct difference to be heard between two reproductions, while at the same time it is difficult, sometimes even impossible, to say

which one is the "best". We seem to have reached a point where the problems are becoming less technical and more artistic, and where we have to consider a gramophone or a radio set no longer as a link between a signal-generator and a voltmeter but as a musical instrument. This will be the theme of J. RODRIGUES DE MIRANDA *). As I see it, our predominant problems concern at present microphone positioning, studio acoustics and sound distribution in the auditorium. Although they may seem to lie outside the scope of Section I they are so essential that we simply have to consider them.

These problems are not all new, but are beginning to create more general interest. This applies especially to *stereophony*, the subject of V. L. JORDAN's contribution *). To explain stereophony it is often suggested that one should place in the studio a "curtain" of microphones, connected to corresponding loudspeakers forming a similar curtain in the auditorium where the sound field is then said to be a copy of the one in the studio. I cannot believe, however, that these two sound fields will even remotely resemble each other, and certainly not when only two or three channels are used as a rough approximation of the curtain.

In my opinion stereophony is based on the remarkable property of the mind to interpret the sound waves, coming from two different sources and differing only in time of arrival and in intensity, as coming from one single phantom source. When one analyses the elements relevant to direction perception in the way K. DE BOER has done, this effect becomes less astonishing. It shows also how one can create stereophonic illusions by simulating these relevant elements in the ears of the listener and why this can be done by means of two loudspeakers. It further follows that the microphones should not be placed at positions corresponding to those of the loudspeakers, as is implied by the curtain theory. They should be placed preferably in an *"artificial head"*, in order not to have to rely on time differences only, which play only a subordinate part in natural listening. The dimensions of this artificial head depend on the circumstances and are not directly connected with those of the human head.

Stereophonic reproduction is certainly a notable improvement, but it does not give a perfect illusion. It widens the hole in the wall to a large window, but it does not place us in the hall where the orchestra is playing. Most marked of all are its deficiencies when the public is applauding. We then realise that the sound is coming from the orchestra and not from all round us. The reverberation suffers from the same defect.

This leads us to the problem of the acoustics of the auditorium and from there back to the acoustics of the studio. Here again and perhaps even more so, we are entering the artistic domain, and have to tread cautiously. Being technical and scientific people, however, we simply have to enter this field; we have to find out what physical properties correlate with the approval of the musician. SABINE's genius gave us the reverberation time but we know that it is not fully adequate to characterize the acoustics of a hall. W. FURRER *) will discuss new concepts, and P. V. BRÜEL *) the instruments to measure them. In Section II H. S. KNOWLES will treat the problem of the artificial imitation of acoustical environments. We ourselves have just started some experiments to analyse room acoustics by reproducing the reverberation artificially. This will enable us to change variables rapidly, and to compare more conveniently their effects. At the same time we hope to find out what is indispensable in creating a satisfactory illusion and so to trace the essential elements that make for good acoustics [1]).

[1]) See report of the Eindhoven excursion, page 300, in this issue.

AUTOMATIC RECORDING OF FREQUENCY IRREGULARITY IN ROOMS

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Summary. To save the tedious job of calculating the frequency irregularity *FI* from a room response curve, two different methods for obtaining direct recording of the *FI* as a function of frequency are developed. In the one method the room response is recorded with a microphone, amplifier and level recorder. On the recorder is fixed a contact strip with 100 silver lamellae which by means of an electronic scaler registers the total "length" of the room response curve in dB. The other method uses the output voltage from the moving coil in the level recorder. This voltage is proportional to the *FI* and is recorded on a second level recorder with extremely low recording speed. The two sets of apparatus have been used for measuring the diffusion effect of three different types of diffusors in a small model room.

Sommaire. Description de 2 méthodes pour enregistrer directement l'irrégularité en fréquence *FI* d'une salle en fonction de la fréquence, de façon à éviter le calcul fastidieux de cette caractéristique à partir de la courbe de réponse de la salle.

a) Cette réponse est reçue par microphone, amplificateur et enregistreur de niveau sonore. Sur l'enregistreur est fixée une bande de contact à 100 lamelles d'argent qui indique la "longueur" totale en dB de la courbe de réponse.

b) On recueille aux bornes de la bobine mobile d'un enregistreur de niveau, une tension proportionnelle au *FI* enregistrée sur un second enregistreur à faible vitesse.

Application: Mesure de l'effet de diffusion produit par 3 modèles de diffuseurs dans une petite maquette de salle.

Zusammenfassung. Es werden zwei Verfahren zur automatischen Aufzeichnung der Frequenz-Irregularität *FI* als Funktion der Frequenz beschrieben, die zur Untersuchung dreier Typen von schallstreuenden Elementen in einem Modellraum benutzt werden. Bei einem Verfahren benötigt man einen Zusatzkontakt am Pegelschreiber, mit dem man bei Aufnahme einer Frequenzkurve des Raumes die „abgewickelte Länge" des Frequenzganges elektronisch zählen kann, während beim zweiten ein weiterer Pegelschreiber die der *FI* proportionale Tauchspulenspannung des ersten aufzeichnet.

In room acoustics the last ten years have shown that an examination of a room's reverberation time is not always sufficient; thus, for example, in respect of acoustic quality some radio studios with approximately the same size and reverberation time have in practice shown themselves to be widely different.

Attempts have therefore been made in recent years [1], [2], [3] to find a measurable quantity other than volume and reverberation time which would characterize a room's acoustics. Particular attention has been paid to the sound field's inhomogeneity, and a tentative quantity for its measurement was defined by BOLT and ROOP, the so-called frequency irregularity *FI*. To estimate a room's *FI*, a loudspeaker with a good frequency response, connected to a beat frequency oscillator, B.F.O., is placed at

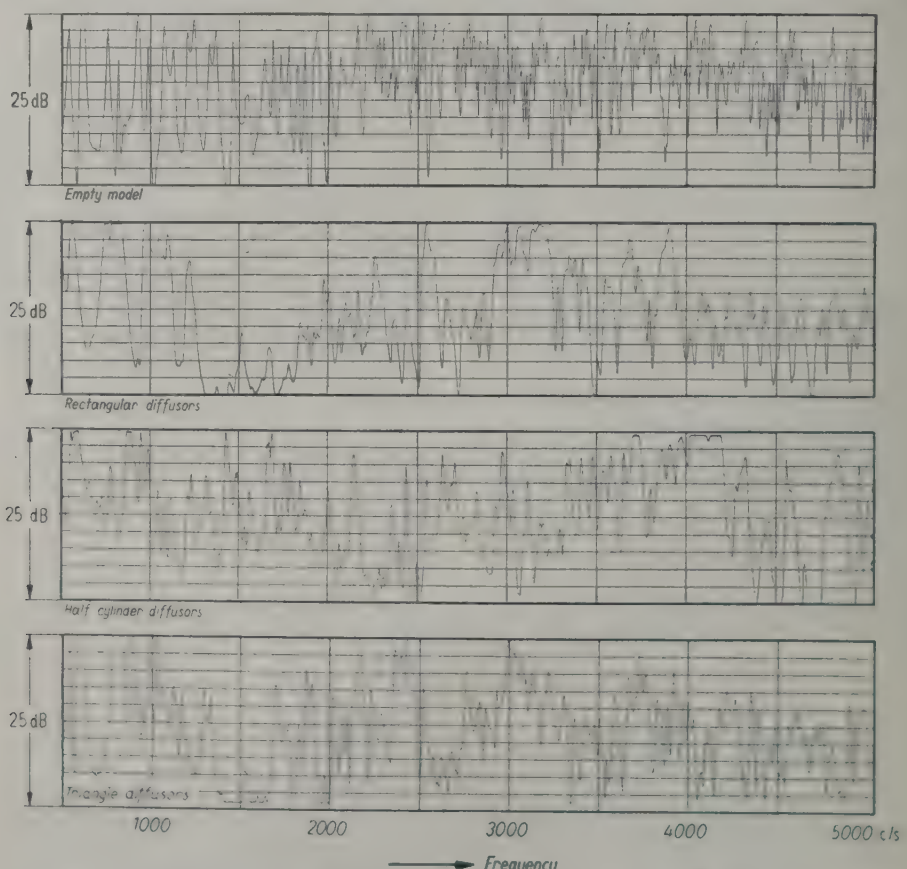


Fig. 1. Frequency response curve in a model room, with different diffusors.
Above: empty room without diffusors.
Below: with diffusors of rectangular, semi-circular and triangular cross-section.

one end of the room, and a pressure microphone, connected to a level recorder via an amplifier, at the other end. The B.F.O. is mechanically coupled to the level recorder, so that the recording paper moves synchronously with the frequency variation, and the frequency scale is made linear by using a circular cut in the B.F.O.'s main condenser plates.

The room's frequency response is recorded with a very slow frequency variation of about 1 ··· 5 c/s per second. A typical set of such curves is reproduced in Fig. 1. BOLT and ROOP define *FI* as the total of the differences in dB between all the maximum and minimum points of the response curve, divided by the frequency band in question. The unit of *FI* is therefore dB per c/s. The figure for smaller solo studios is about 1 ··· 3 dB per c/s, and for studios with bad acoustics, 5 ··· 10 dB per c/s. The value varies with frequency, so *FI* has to be given as a curve, i.e. as a function of frequency.

To record the frequency response of a room is an easy procedure, but to calculate from that curve *FI* as a function of frequency is a big job. This paper describes a procedure for automatically recording *FI* by either of two different methods. The first set-up is shown in Fig. 2.

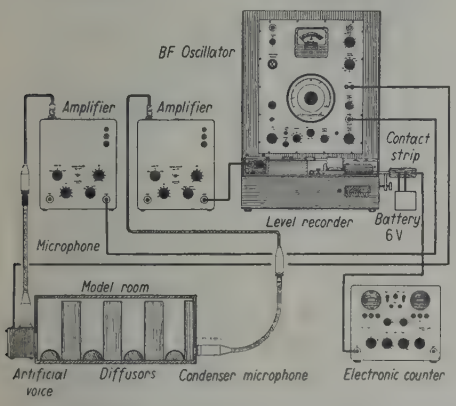


Fig. 2. Measuring set-up with electronic counter and special sound source for the semi-automatic estimation of frequency irregularity.

As test room a small model of a radio studio is used, provided with different types of diffusors (Fig. 3). A normal B.F.O. is converted to give a linear frequency scale in the range 500 ··· 3000 c/s.

As sound source we use an Artificial Voice comprising a highly damped loudspeaker and regulating microphone, which, via a microphone amplifier and the B.F.O.'s compressor, regulates the loudspeaker sound pressure to a constant value. A thick layer of porous material is placed between the Artificial Voice and the room. This material has a high and purely resistive acoustical impedance, so that, as long as the sound pressure is constant on one side

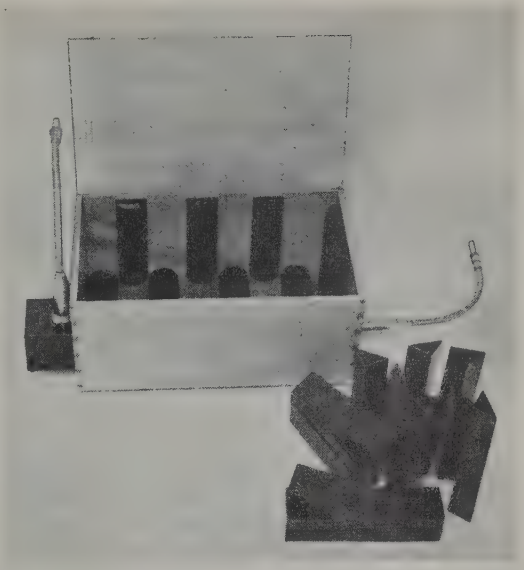


Fig. 3. Photograph of model room and the three types of diffusors used.

of the porous layer, the sound emission on the other side will be constant. Owing to the high acoustical impedance, a variation of the sound pressure on the room side of the layer will not have any influence on the regulating microphone. In other words, the room response being recorded will be unaffected by irregularities in the sound source characteristic. Fig. 4 (top) shows a cross-section of the Artificial Voice provided with the porous material.

At the other end of the model room is placed a condenser microphone connected to a level recorder via an amplifier. The room response is thus recorded as a function of the frequency. The level recorder and the B.F.O. are mechanically connected by means

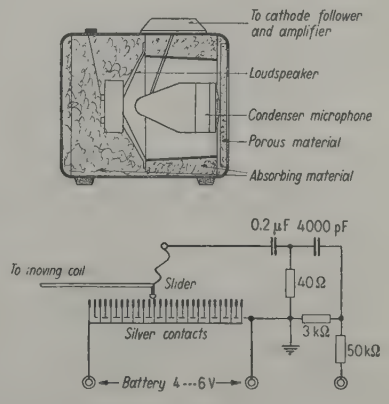


Fig. 4. Cross-section of sound source with flat frequency characteristic and high internal acoustic impedance. Below: principle of the contact arrangement for counting the total length of the frequency curve.

of a chain drive. The curves shown in Fig. 1 are recorded directly from the model with this set-up.

A small contact strip (Fig. 4, below) is mounted on the right-hand side of the recorder, and the recorder's slider is coupled to the slider on this contact strip. Each time the latter slider moves one contact an electrical impulse is obtained which is registered on an electronic counter. In other words, this set-up allows the recording of the total length of the movement of the recorder stylus. The B.F.O. sweep is linear in time, so that if the electronic counter is read at regular periods, say 10 s or $\frac{1}{2}$ min., that is, at constant known frequency intervals, *FI* in dB per c/s can be immediately obtained.

Fig. 5 shows the results from the model room using three different kinds of diffusors. The two sets of curves are for different positions of the diffusors. The points marked with circles are mean values over a wider frequency interval. It is seen that points measured for small frequency intervals fluctuate very much, so that better results are obtained by measuring *FI* over wider intervals. As seen from the curves, the empty room has of course the highest *FI*. In the low frequency range the best diffusors are rectangular in cross section, and the worst, triangular. At high frequencies, the rectangular are not as good as the semi-circular, but are still better that the triangular.

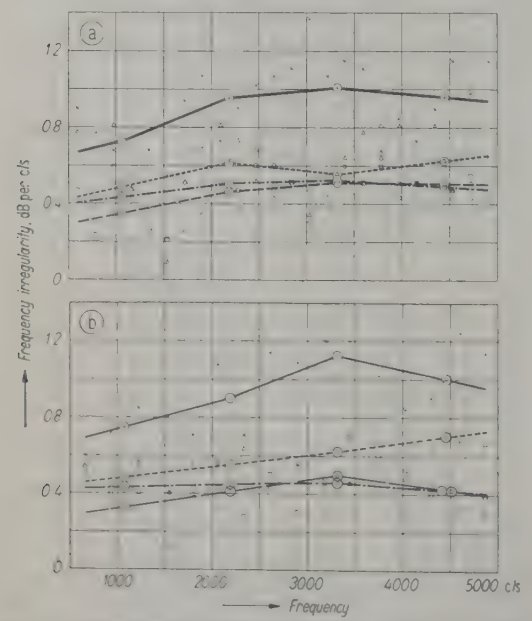


Fig. 5. Experimental results in model room with three different types of diffusors. Curves taken with two positions of the diffusors:
——— the empty room,
- - - - - - - triangular diffusors,
— — — — semi-circular diffusors,
— — — rectangular diffusors.

It should be noted that the frequency irregularity has much lower values measured in a model room than they would have in a full sized studio. The reason is that the reverberation time in a model room is much shorter than in the studio, as the ratio of surface to volume is much greater in the model room. This means that the "quality" of the natural oscillations will be less in the model room, the peaks and valleys will be less pronounced, and as a result a lower value of *FI* will be obtained.

The above method considerably reduces the work of calculation involved in estimating *FI*, but it is actually possible to achieve a completely automatic recording of *FI* as a function of frequency.

Fig. 6 shows the measuring set-up. The sound source, microphone amplifier are just as in Fig. 2. With this fully automatic method, the feed-back voltage from the driving coil in the recorder is led

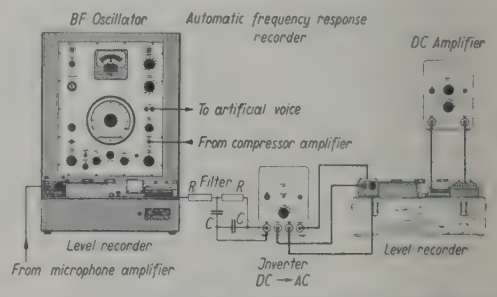


Fig. 6. Principle of the fully automatic set-up for measuring frequency irregularity.

through a filter, to eliminate possible hum, and then fed to a d.c.—a.c. inverter, which converts the slowly varying voltage from the feed-back coil into 400 c/s a.c., which can then be recorded on another level recorder.

The voltage produced by the feed-back winding in the level recorder's driving coil system is proportional to the speed, and therefore, with a constant rate of frequency variation, proportional to *FI* expressed directly in dB per c/s.

However, as is clearly indicated in the curves of Fig. 5, we are not interested in the instantaneous values of *FI*, but on the contrary wish to have them integrated over a relatively broad frequency band. Therefore, in the second level recorder a d.c. amplifier is inverted in its feed-back circuit, whereby the recording speed can be reduced to 1.5 mm/s, which is quite satisfactory for most room-acoustic measurements of *FI*.

The whole apparatus is adjusted by giving the first level recorder a uniform change of level, for example by means of a B.F.O. with a logarithmic scale shifting frequency quite slowly, and with a small condenser in series with the level recorder's potentiometer. The voltage will then vary logarith-

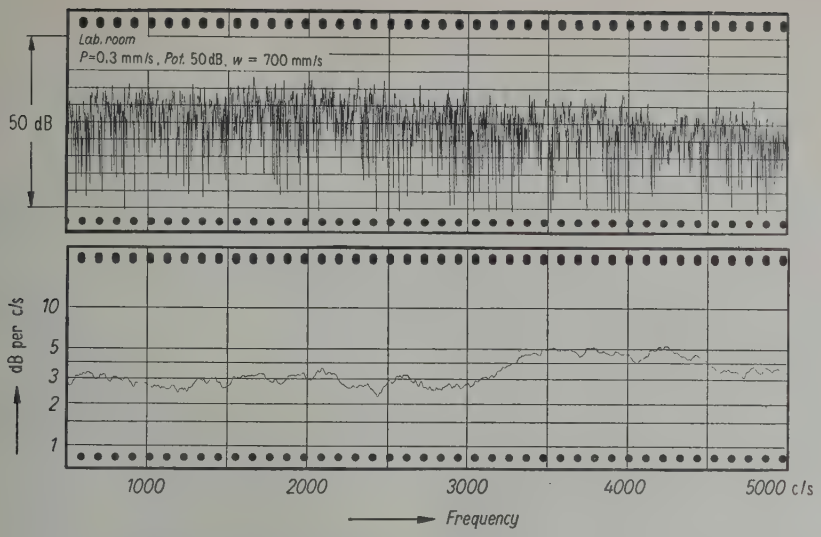


Fig. 7. Frequency response curve and curve of frequency irregularity for a laboratory, recorded with the set-up of Fig. 6.

mically, that is, will give a linear dB scale as a function of time. On the second level recorder a horizontal line should then be recorded. By comparing the slope of the line on the first recorder with the horizontal line on the second, the whole system can be adjusted.

Fig. 7 shows a typical result from an acoustically relatively good room. The frequency range from 500 ··· 5000 c/s is traversed in about 20 minutes.

Fig. 8 shows the automatic recording in the small model room with the three types of diffusors. On the whole, the results agree with those taken by means of the electronic counting method.

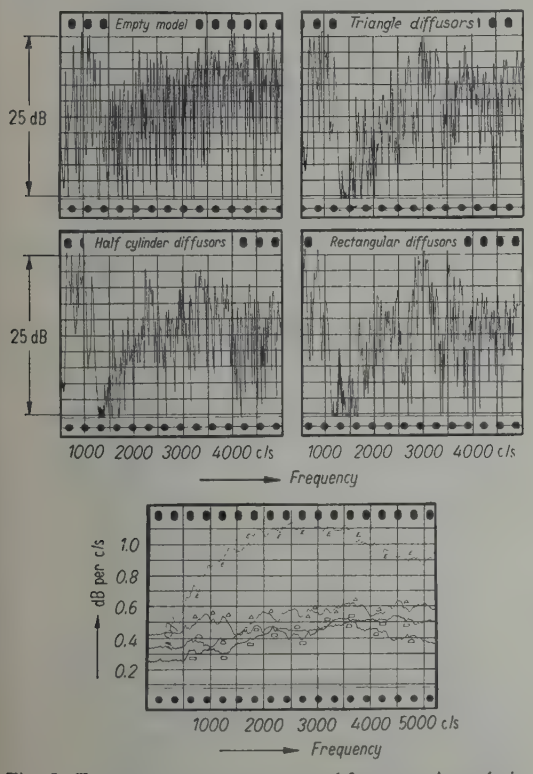


Fig. 8. Frequency response curves and frequency irregularity curves taken in the model room, respectively empty and with the three different diffusors. The frequency irregularity is recorded below on the same curve paper.

References

[1] Bolt, R. H. and Roop, R. W., J. acoust. Soc. Amer. 22 [1950], 280.
[2] Somerville, T. and Ward, F. L., Acustica 1 [1951], 40.
[3] Furrer, W. and Lauber, A., Acustica 2 [1952], 251.

Discussion

Gayford, M. L.: Does the lack of rigidity of the wooden walls of the model room have any effect on the measurements?

Lecturer: It does produce an effect at the lower frequencies used, that is below 500 c/s, but no influence has been found above that frequency.

Kuhl, W.: The measured results from the model room may be explained by the higher absorption of the diffusors compared with the empty room. The diffusors are so small compared with the wavelength that the diffusion effect seems to be very low.

Lecturer: The absorption in the model room was the same with and without diffusors. That is to say that the diffusion effect from the diffusors was real and cannot be explained by differences in the reverberation time.

Jordan, V. L.: Is it not true that the width of the diffusors is more important for the diffusion of sound than the thickness of the diffusors?

Lecturer: I think so, but no measurements have been made for proving that question.

Somerville, T.: I confirm Dr. Brüel's results. In the first issue of "Acustica" we reported such an effect. In this case the slight changes in absorption due to the diffusors were swamped by a constant amount of absorption in the model.

SOME RECENT DEVELOPMENTS IN MAGNETIC RECORDING

by M. Camras

Presented by H. A. Leedy

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Summary. Trends in magnetic recording take advantage of its erasability, fast response, and high resolution. Examples of recent developments are: high-output magnetic tapes, the magnetic modulator head, the "time-compressor", half track recording, and magnetic contact printing.

Sommaire. Les recherches en matière d'enregistrement magnétique portent sur la facilité d'effacement de l'enregistrement, la rapidité de réponse de l'appareil et son grand pouvoir de résolution. Exemples de perfectionnements récents: rubans magnétiques à grand rendement, tête du modulateur magnétique, "compresseur de temps", enregistrement d'une demi-trace, et impression magnétique par contact.

Zusammenfassung. Bei der Weiterentwicklung der magnetischen Aufzeichnung nutzt man die Löschbarkeit, die große Reaktionsgeschwindigkeit und das hohe Auflösungsvermögen aus. Beispiele neuerer Entwicklungen sind: hochempfindliche Bänder, der magnetische Modulationskopf, der „Zeitkompressor", die Doppelspuraufzeichnung und die magnetische Kontaktkopie.

1. Introduction

Two important trends in magnetic recording are notable: 1) better resolution allows good high frequency response at low speeds, 2) unusual applications take advantage of erasability and fast response.

Fig. 1 shows how the speed of the record medium has been reduced since the inception of magnetic recording a half century ago. The danger of extrapolation is illustrated by the dashed portion of the curve, which predicts that by 1956 we shall be pretty much at a standstill. It turns out that this is not pure nonsense, however, because we now have a magnetic head that operates at speeds down to zero.

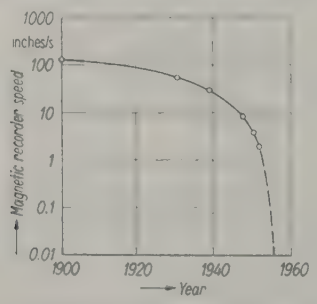


Fig. 1. Trend of typical magnetic recorder speeds.

2. Magnetic tapes

Fig. 2 shows the characteristics of a "high output" tape recently made available. Compared to standard brown oxide tape, the high output tape gives approximately 6 to 8 dB more output for a given distortion level. Frequency response and bias requirements are practically the same as for regular tape, allowing interchangeability. The improved tape gives about 4 dB more output for a given input, and, in addition, allows a 4 dB rise in input level. Most important, the noise level is not increased, so that signal-to-noise ratio is improved.

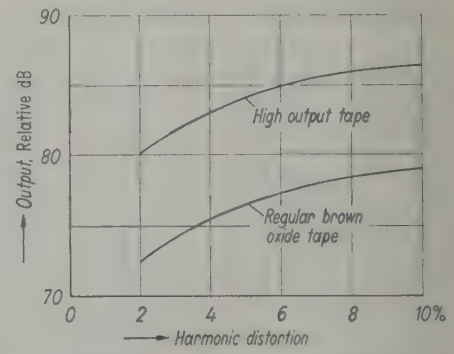


Fig. 2. Comparison of high-output vs. regular tape shows increased output for the same distortion.

Certain applications, such as pulse recording, require tape that is free from "dropouts". Dropouts are blank spots such as might occur from a nonmagnetic bubble in the tape coating. In sound recording an occasional dropout is not detectable. But in electronic computers the loss of a single digit out of millions can cause trouble. Tapes of extreme reliability are now available for such applications, made by a combination of careful manufacturing, testing, and selection.

3. The magnetic modulator playback head

One basic concept in magnetic recording is that the playback voltage tends to be directly proportional to frequency. This rise of 6 dB per octave is caused by pickup heads that are sensitive to *rate of change* of recorded flux, rather than to recorded flux itself. As speed or frequency approach zero, the output of such heads approaches zero.

The magnetic modulator head overcomes these limitations. Its output tends to be constant for all frequencies, for it is sensitive directly to flux. High outputs of more than a volt are obtainable without amplification. The principle of operation is similar to that of a magnetic amplifier. As shown in Fig. 3, an a.c. source feeds a balanced bridge consisting of A, B, R and R'. In balanced condition there is an equal flux a through the outer core legs, and no voltage is present between x and y. A playback flux

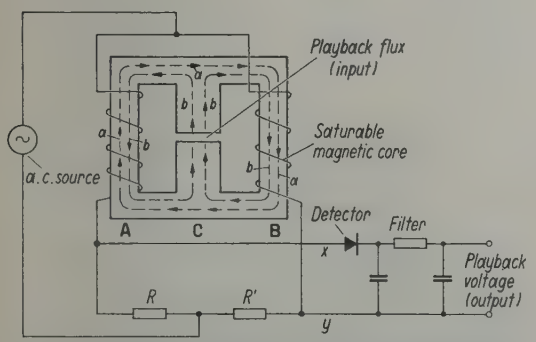


Fig. 3. Principle of the magnetic modulator playback head.

introduced into the centre leg sets up flux b as shown. On the left side of the core, fluxes a and b are opposed so that this leg becomes less saturated. On the right side of the core, fluxes a and b add to make that leg more saturated. The bridge is now unbalanced and a voltage appears between x and y. Similarly for the reverse cycle. When we rectify and filter the bridge output at x, y, we get a steady voltage which is a linear function of the playback flux in the centre leg C.

Some applications of the magnetic modulator head are:

1) pickup at low velocities of a few cm per hour or per day,
2) true reproduction of complex recorded waves,
3) detailed examination of recorded data, allowing stoppage at any point,
4) low-cost systems requiring little or no amplification.

4. The "time compressor"

To fit an exact time schedule, a recorded programme is often played back more slowly or faster. Experience shows that a two or three per cent difference is understandable but, because the pitch is changed, the voice does not sound natural. A ten per cent difference is unpleasant and difficult to understand.

FAIRBANKS, EVERITT, and JAEGER [1] have developed a "time compressor" that reduces playback time *without changing the pitch*. Compressions of 10 to 20 per cent are hardly noticed and even 50% is feasible. The principle of operation is that, human speech being so redundant, we can cut out 90% of a sentence, at very small intervals, and still make sense out of what is left. An analogy is shown in Fig. 4a, where a sign is painted on a board fence (left). When we remove every second board and put the rest together the sign remains readable (right). Preliminary experiments were made by cutting out small bits of tape from a recorded programme and pasting the remainder together.

The machine of Fig. 4b does this automatically. Original sound is first recorded on a loop of tape. This recording is picked up by four heads mounted on a rotating drum. For a two-to-one compression, the drum surface moves in the same direction as the tape, but only half as fast. This means that when head 1 reaches position 2, points A, B, C, and D on the tape have reached A′, B′, C′, and D′ respectively. During this time head 1 has been scanning the interval AB on the tape. Next, head 4 is in position. It scans from C to D. We notice that the portion BC of the tape is not scanned at all. Similarly we divide the rest of the recording into equal intervals; half are picked up, and half are discarded.

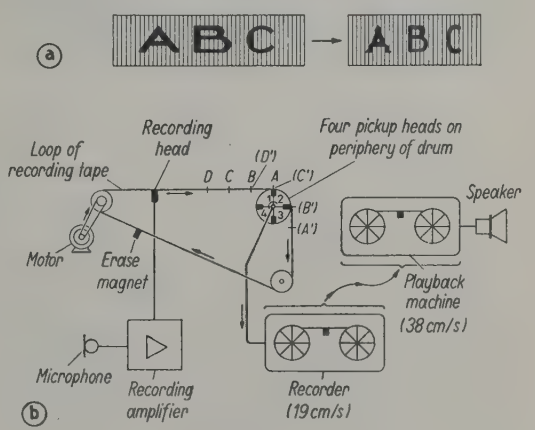


Fig. 4. The "time compressor":
a) Analogy of sign painted on board fence. Discarding every other board putting remainder together still leaves it readable.
b) Apparatus for accomplishing time compression.

Since the relative motion between tape and scanning head is only half the loop speed, the pickup frequency fed into the 19 cm/s recorder is half that of the original sound frequency. The tape made at 19 cm/s is finally played back at 38 cm/s, restoring the original frequency.

We can use the same machine also as a time expander, by reversing the scanning drum so that it increases the redundancy of speech. A compressed or expanded voice sounds natural, but the subject

just "speaks faster or more slowly". Applications of the time compressor are: reduction of bandwidth required for audio transmission, new musical effects, and saving of expensive time purchased on broadcast networks. As Dr. EVERITT remarked: "A politician who has nothing to say, can now say it in half the time."

5. International standardization of playback characteristic

There are almost as many recording frequency characteristics as there are companies who make recorders. To facilitate international exchange of tape records, the C.C.I.R. Study Group X meeting in Geneva in 1952 recommended a "standard replay chain" with a characteristic as shown in Fig. 5. This response is typical of systems where high frequencies are equalized entirely during recording, and low frequencies entirely during playback.

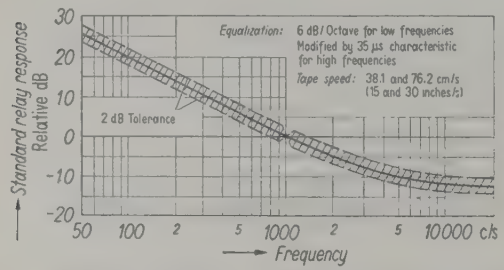


Fig. 5. Playback response for magnetic tape recorders. CCIR study group X recommendations for std. replay chain (equalization for head characteristics to be added).

The main problem lies in compensating for the playback head and input transformer, which are unknown and variable links in the "standard replay chain". One solution is to make available a calibrated tape, with a series of test frequencies that play back flat on a correctly adjusted machine. Mr. DAVIES of the BBC is circulating a tape of this kind among laboratories and manufacturers to check its calibration.

6. Half-track recording for sound-on-film

Magnetic sound on 16 mm film is widely used for industrial, educational and amateur movies. As shown in Fig. 6a, the SMPTE standard calls for a magnetic strip, 2.5 mm (0.10 inch) wide. A modification of this standard is a magnetic track of half-width on a film which is printed with a photographic sound track, as in Fig. 6b. The same projector allows selection of magnetic or photographic track.

In Fig. 6b the photographic track contains permanent sound, while the magnetic track may carry the comments of a teacher, or a sales message to

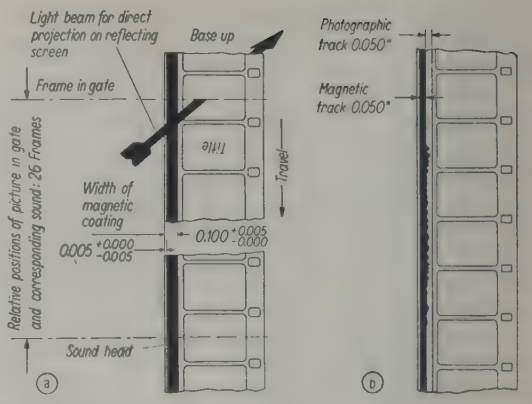


Fig. 6. Magnetic sound films:
a) Proposed American standard for 16 mm magnetic film (J. Soc. Mot. Pict. Telev. Engrs. July 1951).
b) Half track magnetic strip, plus optical sound.

a particular audience. The magnetic sound is, of course, erasable at will. An important use is for translations into foreign languages and dialects. The translator listens to the photographic sound track; then records the translation on to the magnetic track. This method is so simple that it allows showing in native languages that were not economically possible before.

7. Duplication by Magnetic Contact Printing

Duplicating magnetic records by "dubbing" or re-recording is generally slow and expensive. Fig. 7 shows a magnetic contact printing machine which prints several blank tapes at once. The master record is of high coercive magnetic material which is unaffected by the process. It is brought into contact with ordinary blank tape, and while in contact, an a.c. field is applied and reduced to zero. Copies are of high quality and maximum recording level.

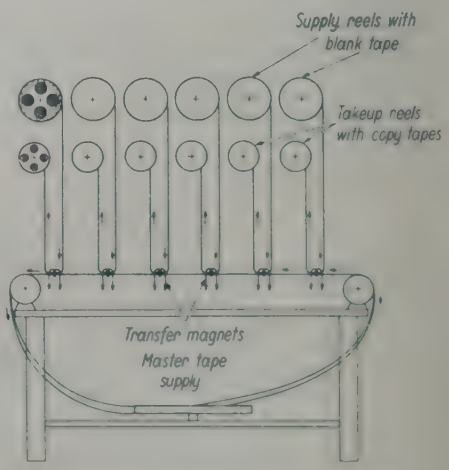


Fig. 7. Schematic layout of multiple duplicator.

8. Conclusion

We have described a few interesting and important developments in magnetic recording, but many others could be mentioned. Some of these are: binaural and stereophonic recorders, now extensively used; ferrite heads which are excellent at high frequencies and so hard as to be virtually wear-proof; turn-in-gap and multiple-gap-heads for efficient erasure; the cross-field head which gives a very sharp recording field by combining a vertical field with the usual semi-circular field at the gap; memory devices for electronic computers; and artificial reverberation generators.

References

[1] Fairbanks, G., Everitt, W. L. and Jaeger, R. P., Method for time or frequency compression-expansion of speech. Proc. Inst. Radio Engrs. [1953], (unpublished).

See also:

Camras, M. and Herr, R., Duplicating magnetic tape by contact printing. Electronics **22** [1949], No. 12, 78—83.

Lueck, L. B. and Wetzel, W. W., Performance of high output magnetic tape. Electronics **26** [1953], No. 3, 131.

Wiegand, D. E., A flux sensitive head for magnetic recording play back. Proc. Inst. Radio Engrs. [1953], (unpublished).

Discussion

Begun, S. J.: I would like to make some supplementary remarks about some interesting developments in the field of magnetic recording. A reproducing head has been built which employs an electron beam tube as a transducer. This head, similar to the one mentioned by Dr. Leedy, is responsive to the amplitude of the recorded flux.

Another development I would like to call to your attention is that of embedding magnetic particles in a rubber base. A recording medium of this type permits high-speed operation while the head is in contact with it. This is particularly important when pulses need to be recorded. With the head in physical contact with the medium, more pulses can be packed per linear unit length of the medium. It might be worth mentioning that considerable advances have been made in the quantity production of narrow-gap heads with great physical stability. These heads which use resin as a medium for holding the pole pieces in proper relationship can now be provided at reasonable prices.

Schiesser, H. O.: Time compression systems, using the principle mentioned in Mr. Camras' paper, were designed by Schüller in 1938 and built in about 1000 sets. A compression- as well as extension-ratio of 1 : 5 was maintained.

Mercer, D. M. A.: In recording aircraft jet noise for analysis later, how accurate is the overall process likely to be?

Lecturer: The results depend on the type of recorder, the tape used, etc. Under best conditions, using standardizing techniques, we can come within the $\pm$ 1 dB specification.

Baerle, F. C. G. van: Is the rotating type head assembly in the time expansion device similar to the "Edispot" rotating head assembly, described a few years ago?

Lecturer: The "Edispot" as I understand it, is a movable head that enables the operator to listen repeatedly to a short length of tape, while the tape recorder is stopped or is being moved slowly by hand. It enables an operator to locate accurately the points for cutting and editing. I do not think the design is suitable for a "speech compressor".

DIE RAUMAKUSTISCHE DIFFUSITÄT IN SCHALLAUFNAHME- UND RADIOSTUDIOS

von W. Furrer und A. Lauber

P.T.T. Bern

Zusammenfassung. Die Homogenität eines Schallfeldes in einem Raum, auch Diffusität genannt, ist besonders für Schallaufnahme- und Radiostudios wichtig. Es wurde schon früher gezeigt, daß die „mittlere Höhe eines Berges" des Raumfrequenzganges ein praktisch brauchbares Maß für die Diffusität ist. Die direkte Ausmessung solcher Frequenzgänge ist jedoch mühsam, so daß eine Meßapparatur entwickelt wurde, mit der die so definierte Diffusität verhältnismäßig leicht und rasch gemessen werden kann.

Summary. The homogeneity of a sound field in a room, which is called "diffuseness", plays an important part in radio and sound recording studios. Earlier investigations have already shown that the "mean height of a peak" of the frequency response of a room is a suitable index of diffuseness. The determination of this quantity from the frequency response is, however, so laborious that a special measuring apparatus had to be developed, ensuring quick and easy measurement.

Sommaire. L'homogénéité du champ acoustique dans un local, qu'on peut encore appeler "diffusivité", joue un rôle important dans les studios de radio et d'enregistrement. On a pu montrer auparavant que la "hauteur moyenne d'un sommet" de la courbe de fréquence du local est un paramètre définissant la diffusivité. Il est cependant très laborieux de déterminer ce facteur à partir de la courbe de fréquence; pour cette raison on a développé un nouvel appareil qui permet une mesure facile et rapide de la diffusivité.

Es ist heute allgemein bekannt, daß außer den klassischen raumakustischen Parametern, deren wichtigster die Nachhallzeit ist, noch andere Faktoren das akustische Verhalten eines Raumes maßgebend beeinflussen. So hat es sich vor allem gezeigt, daß die in einem Raum vorhandene Diffusität

besonders in Schallaufnahme- und Radiostudios, aber auch ganz allgemein in allen Räumen, eine entscheidende Rolle spielen kann. In einer früheren Arbeit [1] wurde gezeigt, daß die „mittlere Höhe eines Berges" des im stationären Zustand gemessenen Raumfrequenzganges eine Größe ist, die die Homogenität des Schallfeldes und damit die Diffusität unmittelbar beschreibt. Als Frequenzgangschwankung F_ν wird dabei definiert:

$$F_\nu = \frac{\Sigma p_{max} - \Sigma p_{min}}{\Delta \nu}.$$

Dabei bedeuten

Σp_{max} Summe der Schalldruckmaxima („Berge"),
Σp_{min} Summe der Schalldruckminima („Täler"),
$\Delta \nu$ Frequenzintervall.

Daraus bestimmt sich die Diffusität D_ν zu:

$$D_\nu = F_\nu / n,$$

wobei n die Anzahl der Schalldruckmaxima ist.

Diese Methode wurde inzwischen systematisch für die Untersuchung und Dimensionierung von Schallaufnahme- und Radiostudios verwendet, und es konnten mit ihr Erfahrungen gesammelt werden, die die Zweckmäßigkeit dieser so definierten Diffusität und ihren Einfluß auf das akustische Verhalten eindeutig belegen. Dabei mußte zunächst die Meßmethode ausgebaut und rationalisiert werden, da die vollständige Auswertung von registrierten Frequenzgängen so mühsam und zeitraubend ist, daß sie die praktische Anwendung der neuen Methode in Frage stellt. Zu diesem Zwecke mußten nicht nur auf der Sendeseite, sondern besonders auch auf der Empfangsseite die Meßapparaturen ergänzt und vervollständigt werden.

In der erwähnten Arbeit ist bereits beschrieben, daß als quasi-punktförmige Schallquelle das Antriebssystem eines Trichterlautsprechers mit einer verhältnismäßig großen mechanischen Impedanz verwendet wird; ferner wird der Abstand d zwischen Lautsprecher und Mikrophon immer zu $d = \sqrt{0{,}1\,A}$ gewählt, wobei A die mittlere Raumabsorption darstellt. Die beiden ausgemessenen Frequenzbereiche, die für das akustische Verhalten eines Raumes besonders charakteristisch sind, wurden gegenüber den früheren Untersuchungen leicht modifiziert; in der folgenden Tabelle sind sie angeführt.

Tieftonbereich:	250···300 Hz	
	300···350 Hz	
	350···400 Hz	
	400···450 Hz	Mittelwert 425 Hz
	450···500 Hz	
	500···550 Hz	
	550···600 Hz	
Hochtonbereich:	1300···1400 Hz	
	1400···1500 Hz	
	1500···1600 Hz	
	1600···1700 Hz	Mittelwert 1650 Hz
	1700···1800 Hz	
	1800···1900 Hz	
	1900···2000 Hz	

Das Registrieren von genauen und reproduzierbaren Frequenzgängen erfordert einen sehr stabilen Generator mit einem genauen Frequenzablauf und ganz präzisen Frequenzmarkierungen. Die Geschwindigkeit des Frequenzablaufs ist abhängig von den akustischen Eigenschaften des Raumes und von der Schreibgeschwindigkeit des Registriergerätes; sie muß im allgemeinen in der Größenordnung von 2 Hz/s liegen. Das apparative Kernstück der neuen Methode ist der auf der Empfangsseite verwendete mechanische Integrator, der die Summe der „Steigungen", die Summe der „Gefälle", die Anzahl der „Berge" sowie die Anzahl der „Täler" direkt abzulesen gestattet. Dieses Gerät zeigt Abb. 1. Es besteht im Prinzip aus zwei Freiläufen, wie sie in ähnlicher Form bei Fahrrädern bestehen, mit denen die Bewegungen des Zeigers in beiden Richtungen summiert werden. Zudem wird bei jeder Bewegungsumkehr ein Kontakt betätigt, der einen Impuls für einen Zähler liefert, wodurch sowohl die Anzahl der „Täler" als auch die Anzahl der „Berge" gezählt wird.

Der ursprüngliche Gedanke war der, diesen mechanischen Integrator direkt durch den Pegelschnellschreiber anzutreiben. Es hat sich jedoch gezeigt, daß besonders bei steilen Bergen oder sehr tiefen engen Tälern die zur Verfügung stehende Kraft nicht mehr ausreicht, um die Reibungen des Integrators vollständig zu überwinden, wodurch sich die Genauigkeit vermindert. Daher wird der Integrator durch einen manuell betätigten Folgezeiger angetrieben (Abb. 2), ein Verfahren, das allerdings etwas mehr Zeit beansprucht, aber gegenüber der direkten Ausmessung immer noch eine sehr große Zeitersparnis ergibt.

Die beschriebene Methode wird am Beispiel eines Konzertstudios von 5000 m³ illustriert, das umgebaut wurde, um die im ursprünglichen Zustand ungenügende Diffusität zu erhöhen. Die Abb. 3 zeigt den Raum im ursprünglichen, die Abb. 4 im umgebauten Zustand. In der folgenden Tabelle sind die vor und nach dem Umbau gemessenen Diffusitäts-

	D_ν	
	Tieftonbereich (425 Hz)	Hochtonbereich (1650 Hz)
vor Umbau	0,170	0,089
nach Umbau	0,108	0,087

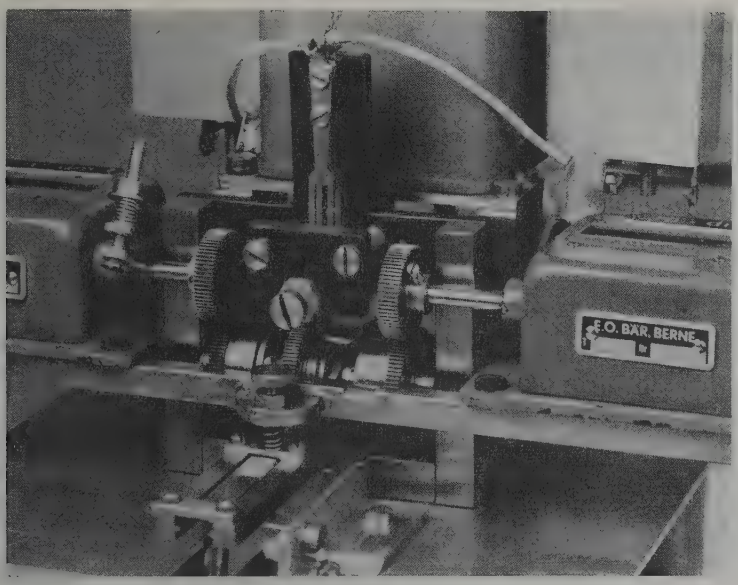


Abb. 1. Mechanischer Integrator zur Auswertung von Raumfrequenzgängen; links und rechts die beiden Zähler der „Steigungen" und „Gefälle".

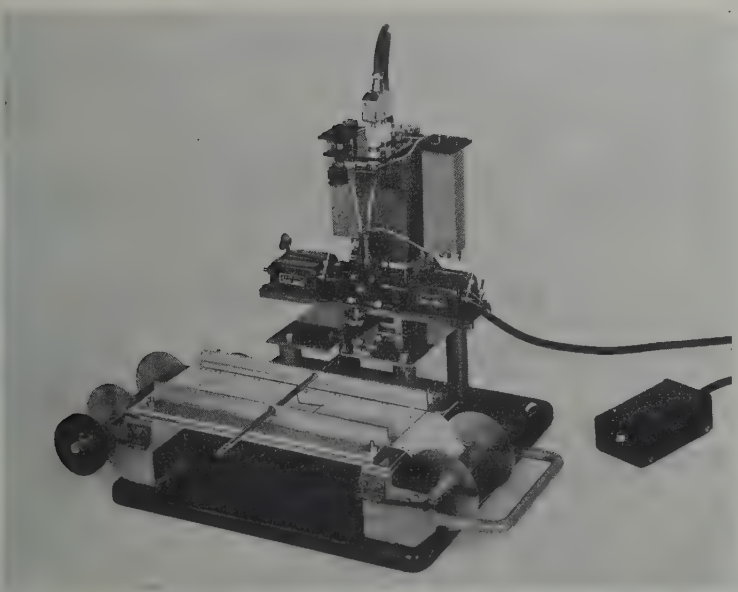


Abb. 2. Integrator durch Folgezeiger angetrieben für die Auswertung eines Registrierstreifens.



Abb. 3. Konzertstudio von 5000 m³ im ursprünglichen Zustand mit wenig Diffusität.



Abb. 4. Umgebautes Konzertstudio (vgl. Abb. 3) mit vergrößerter Diffusität.

werte zusammengestellt; dabei ist zu beachten, daß einer guten Diffusität ein *kleiner* Wert von D_v entspricht.

Die Werte zeigen sehr deutlich, daß sich die Diffusität im Tieftonbereich stark verbessert hat, während sie im Hochtonbereich, wo sie schon vor dem Umbau groß war, praktisch unverändert geblieben ist. Die bei Orchesteraufnahmen erzielten Resultate stehen durchaus im Einklang damit; die Qualität ist heute ausgezeichnet, obschon sich die Nachhallkurve nur unwesentlich geändert hat.

Die neue Methode liefert also ein Kriterium, das den Einfluß der Raum*form* auf das akustische Verhalten erfaßt und sich in sehr befriedigender Weise mit den subjektiven Feststellungen deckt.

Schrifttum

[1] Furrer, W. und Lauber, A., Die Diffusion in der Raumakustik. Acustica **2** [1952], 251.

Diskussion

Hunt, F. V.: While the frequency irregularity may be, and probably is, related to the property of "diffuseness" of sound in a room, they are certainly not the same. It may therefore be a mistake to encourage confusion by applying the term "diffusion" to quantities derived directly from the frequency irregularity.

Vortragender: This is true and a more appropriate designation has to be found; may be that "homogeneity" is a better word.

Knowles, H.: These expressions can be tested by the "diffusion" for the free field (asymptotic) case. In this case the loudspeaker irregularities alone can lead to numerical values lying in the room range. This emphasizes the need for care in applying the term "diffusion" to the defined quantity.

Vortragender: It is in any case necessary that the frequency irregularities of the loudspeaker are very much inferior to those of the room; it is easy to satisfy this condition.

Jordan, V. L.: What method of subjective assessment has been used?

Vortragender: The subjective assessment of the quality of a room is a very difficult task and has to be done on a wide statistical base. For the studios in question only the experiences of the responsible sound engineers over a considerable time have been used.

STYLUS-GROOVE RELATIONS IN THE PHONOGRAPH PLAYBACK PROCESS

by F. V. Hunt

Harvard University, Cambridge, Mass., U.S.A.

Summary. The motion of a phonograph pickup stylus in a lateral-cut record groove is analysed with due regard for the first-order effects of elastic deformation of the groove walls. The relative response at any frequency is governed in part by a groove-pickup response function embodying the effect of resonance between the compliance of the groove and the effective mass of the stylus. A translation-loss function further controls the response at fundamental frequency only and imposes a sharp cut-off at a recorded wavelength which is a function of the stylus and record parameters. The distortion response at harmonic frequencies is effectively controlled by a scanning-loss function, which also varies with the recorded wavelength and which imposes its own cut-off at a somewhat shorter wavelength. Experimental measurements with special test recordings confirm the theoretical predictions.

Sommaire. Analyse du mouvement de l'aiguille d'un pick-up dans le sillon d'un enregistrement à coupe latérale, en tenant compte des effets du premier ordre dûs à la déformation élastique des parois du sillon. La réponse relative à une fréquence donnée dépend en partie de l'effet de résonance entre la raideur du sillon et la masse effective de l'aiguille; de plus, une perte translative impose une coupure nette pour une longueur d'onde fonction des caractéristiques de l'aiguille et de l'enregistrement. La réponse en distorsion aux harmoniques est conditionnée par la perte par balayage, variable avec la longueur d'onde enregistrée, et qui impose sa coupure à une longueur d'onde un peu plus courte que la précédente. Les mesures expérimentales confirment les prévisions théoriques.

Zusammenfassung. Die Bewegung einer Tonabnehmernadel in einer Schallplattenrille (Seitenschrift) wird unter Berücksichtigung der Deformationseffekte der Rillenwandungen analysiert. Die relative Charakteristik wird teilweise durch Resonanzeffekte beherrscht, die durch das Zusammenwirken der Rillenfederung mit der effektiven Masse der Nadel bedingt sind. Außerdem wird noch ein nur bei der Grundfrequenz auftretender Übertragungsverlust wirksam, der eine scharfe untere Grenzfrequenz zur Folge hat, die von der Nadel und den Plattenparametern abhängt. Die durch die geometrischen Abtastverhältnisse bedingten harmonischen Verzerrungen sind ebenfalls frequenzabhängig und haben eine etwas höher liegende Grenzfrequenz. Experimente mit Testplatten bestätigten die theoretischen Vorausberechnungen.

The kinematical problem of finding the motion of a spherical stylus that is tracing the modulated groove of a lateral-cut record has been studied [1], previously under the assumption that the groove walls are ideally rigid. A solution has now been obtained [2] for the corresponding dynamical problem in which the first-order effects of the deformation of the groove walls are taken into account.

The dynamical equations governing the stylus motion are formulated in a transformed coordinate system whose origin follows a locus representing the solution of the rigid-wall tracing distortion problem. The elastic deformation problem is then set up in these transformed coordinates, the normal forces acting across the surfaces of contact being just equal to the constraint forces required to give the stylus its assumed motion. Knowing the rigid-wall trajectory allows these constraint forces to be expressed in series form. When the other terms of the dynamical equation are also expressed as Fourier series, the actual stylus motion can be obtained in terms of a similar Fourier series in which the coefficient of each harmonic appears as a power series in the modulation amplitude. The convergence of these power series is such that only the leading term needs to be retained for any combination of parameters for which moderate values of tracing distortion would be predicted by the rigid-wall theory. The relative stylus response can then be expressed as a product of dimensionless factors that are relatively simple functions of the physical parameters of the record and stylus system.

The relative stylus response at the fundamental frequency is given by the product of a groove-pickup response function H and a translation-loss function G. The groove-pickup response function H is displayed in Fig. 1, which is a plot of the expression

$$H = H\left(\frac{\omega}{\omega_0}, \varepsilon\right) = \left\{\left[1 - \left(\frac{\omega}{\omega_0}\right)^2\right]^2 + \varepsilon^2\left(\frac{\omega}{\omega_0}\right)^2\right\}^{-1/2}, \quad (1)$$

ε being a damping parameter and ω_0 the angular frequency at which resonance occurs between the groove wall compliance and the effective mass m of the stylus system referred to the point of contact. Care must be taken to evaluate m at a frequency near ω_0 since incipient mechanical resonances in the stylus or its supporting structure can cause the effective mass to be significantly higher than the value measured statically or at a low frequency. The physical parameters of the stylus and the record material determine the stylus-groove resonance frequency through the relation

$$\omega_0 = \frac{1.27}{m^{1/3}}\left[F_z R_1\left(\frac{E}{1-\nu^2}\right)^2\right]^{1/6}, \quad (2)$$

in which R_1 is the stylus tip radius, F_z is the vertical bearing force, and E and ν are the elastic modulus and Poisson's ratio for the record material. Since

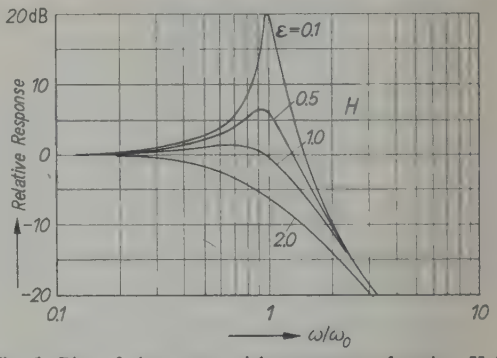


Fig. 1. Plot of the groove-pickup response function H for different values of the damping parameter ε.

the response is only uniform for frequencies well below ω_0, it is obviously desirable to choose the physical constants so that ω_0 exceeds the highest frequency to be reproduced. To satisfy this condition, with typical record materials and stylus dimensions, requires that m be not more than a few milligrams. Unfortunately this condition is not satisfied by most pickups now available commercially.

The translation loss factor G is given (see Fig. 2) by

$$G = 1 - \left(\frac{\omega}{\omega_c}\right)^2 - \left(\frac{\omega_a}{\omega_0}\right)^2 \approx G\left(\frac{\omega}{\omega_c}\right) = 1 - \left(\frac{\lambda_c}{\lambda}\right)^2, \quad (3)$$

in which ω_a is the angular frequency of "free" resonance of the stylus vibratory system when not

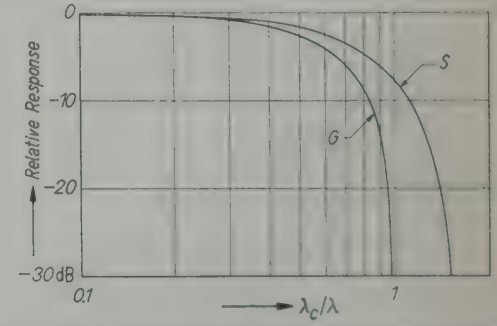


Fig. 2. Plot of the translation loss function G, which controls response at fundamental frequency; and of the scanning loss function S, which is a factor in the response for all frequency components.

in contact with the record, and ω_c is a cut-off angular frequency determined by the linear groove velocity u and the same system parameters defined above; thus,

$$\omega_c = 3.33\, u\left[\frac{E}{R_1 F_z (1-\nu^2)}\right]^{1/3}. \quad (4)$$

The term $(\omega_a/\omega_0)^2$ is ordinarily small for a well-

designed pickup, as suggested by the approximation indicated on the right in eq. (3). Since ω_c varies directly with the groove velocity, eq. (4) indicates that the translation loss is characterized by a cut-off wavelength that is a function only of the stylus radius, the vertical bearing force, and the elastic properties of the record material.

The relative distortion response of the stylus at the second and all higher harmonic frequencies is given as the product of the same groove-pickup response function H and a new scanning loss function S. The translation loss function G does *not* enter as a factor in the distortion output. The scanning loss has its physical origin in nonuniformity of the normal force acting on different portions of the surface of contact, the nonuniformity being caused by variations in curvature of the groove wall due either to the recorded modulation or to the components of motion representing the tracing distortion. Under conditions for which modest tracing distortion is predicted, the variable component of the deformation force is small enough to allow the scanning loss to be expressed in the form of a series in powers of the ratio of 2δ, the diameter of the surface of contact, to the recorded wavelength. Evaluating the diameter of the contact surface then reveals that the scanning loss function can be expressed in terms of the same cut-off wavelength that appears in the translation loss function, as shown by the second curve of Fig. 2 and by the following series:

$$S'\left(\frac{2\delta}{\lambda}\right) = S\left(\frac{\lambda_c}{\lambda}\right) = 1 - \frac{3}{4}\left(\frac{\lambda_c}{\lambda}\right)^2 + \frac{3}{16}\left(\frac{\lambda_c}{\lambda}\right)^4 - \\ - \frac{3}{128}\left(\frac{\lambda_c}{\lambda}\right)^6 + \frac{9}{5120}\left(\frac{\lambda_c}{\lambda}\right)^8 - \ldots \quad (5)$$

The physical factors that give rise to the scanning loss come into play at the fundamental frequency as well as at harmonic frequencies, but since G introduces a substantial loss before S has departed appreciably from unity, the scanning loss is usually ignored when considering the response at fundamental frequency.

A series of test pressings in different record materials were prepared with the recorded signals programmed so that the various response functions could be isolated for direct measurement. Recorded square waves offered a further opportunity for qualitative observations of the influence of various parameters on the transient response. The predictions of the theoretical analysis were substantially verified in all important respects.

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Discussion

Begun, S. J.: I think it might be of interest to direct the attention of the audience to some early work by Otto Kornei, who studied also the problem of translation loss. Kornei also used the Hertz' deformation formula and obtained results which explained the change of frequency response from the outside to the inside groove. Dr Hunt's discussion goes, however, much further than Kornei's work, who limited his investigations to the turning points of the modulated groove.

Lecturer: I am glad that Mr. Begun gave me this opportunity to explain the relation between our work and Kornei's. Dr Miller, who carried through the analysis I have reported, studied Kornei's work carefully and found that he had made one error in applying the Hertz' analysis. This error involved using the groove-wall curvature corresponding to the actual stylus motion rather than that associated with the recorded modulation. When this error is corrected, Kornei's results fall into perfect agreement with the present analysis and represent a special case of Miller's more general treatment (see [3]).

Korn, T.: Has the lecturer examined the influence of the groove elasticity on the phenomenon of the "counter-resonance" formed by the compliance of the stylus suspension and the mass of the arm? We think the influence of the elasticity of the groove in this case seems to be even more important, while the input impedance of the stylus seen from the groove is very large.

Lecturer: I think the effective centre of oscillation would be in a different place for the two resonances in question. For the stylus-groove resonance, oscillation takes place about a dynamic centre conjugate to the stylus tip, and the groove-wall compliance is controlling. For the low-frequency "arm resonance", oscillation takes place about the stylus tip as a centre, and the motion would be relatively uninfluenced by the groove-wall compliance.

Jaquess, D. G.: Reference was made to a very quick method of measuring pickup resonances, that is, by examining the surface noise spectrum with a panoramic type of wave analyser. For the point-groove wall resonance, a record with unmodulated grooves is used. For the free resonance the pickup is allowed to run on the uncut surface, and the assumption is made that there is negligible indentation of the surface.

Lecturer: Undoubtedly some kind of resonance would be observed with the stylus riding on an uncut surface, but I don't think it would be the "free" resonance that I defined. Fortunately the free resonance is not very important anyway, so it probably doesn't make much difference. However, I am glad to hear that someone has actually tried this method of observing the resonances.

Schlegel, F.: We have recorded some frequencies at inner and outer grooves and pressed them from the same master in different materials. There is a good dynamic pickup available in Denmark with both 1 and 4 mg mass on the needle point, but both systems of identical construction. Thus, we have got some measurements of interest concerning the resonance between elasticity of the groove and the mass on the tip. This resonance rises about one octave with a fourth of the mass.

Lecturer: I am glad that our prediction about the shift of the stylus-groove resonance is confirmed. I am especially glad to hear that at least one pickup is now available with an effective stylus mass in the 1 milligram range.

A SYSTEM FOR STEREOPHONIC REPRODUCTION

by V. L. JORDAN

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Summary. When listening to two loudspeakers transmitting the same sound signal, the relation between the angular displacement of the virtual sound source and the difference in intensity level of the two sources is linear within a certain range, and a mean value of the slope of the curve may be fixed at 0.5 dB per degree of angle. This implies the use of microphones with suitable directional characteristics. The stereophonic impression of a two channel system arranged according to this principle is rather satisfactory, although the level is a little too low for sounds originating close to the axis of symmetry.

Sommaire. Dans l'écoute de deux haut-parleurs émettant le même signal sonore, il y a, dans un certain domaine, une relation linéaire entre l'angle de la source sonore "artificielle" et la différence des niveaux de pression sonore: environ 0,5 dB par degré d'angle. L'impression stéréophonique obtenue avec un système à 2 voies conçu suivant ce principe est plutôt satisfaisante, bien que le niveau sonore soit un peu trop faible pour les sons issus des points voisins de l'axe de symétrie.

Zusammenfassung. Bei Wiedergabe des gleichen Schallsignals aus zwei Lautsprechern besteht in einem gewissen Bereich eine lineare Beziehung zwischen der Winkelverschiebung der virtuellen Schallquelle und der Pegeldifferenz der zwei Lautsprecher (Mittelwert der Kurve etwa 0,5 dB/Grad). Eine Zweikanalübertragung nach diesem System mit Mikrophonen passender Richtcharakteristik war ziemlich befriedigend.

Stereophonic reproduction has not hitherto found widespread application, but it seems certain that the trend towards better quality in sound reproduction must lead to the use of stereophony both in cinemas, in theatres with sound reinforcement, over the wireless as well as by reproduction of music in general.

Different arrangements, more or less casually designed, have been applied for stereophonic reproduction, but only a few of them have been based upon a careful study of the facts which are of importance if you wish to obtain a real stereophonic impression. These are the facts concerning the human capacity of sound localization. Let us here limit ourselves to a discussion of the lateral localization in a horizontal plane, since this is obviously most important when dealing with stereophony. The lateral localization of a single sound source is due to differences in the intensity of the sound at the two ears as well as to differences in the time of arrival of the sound signal to the two ears. KIETZ [1] has recently published a review of these facts. It seems to be established that at frequencies below 800 c/s the capacity of localization is due exclusively to time differences, whereas at frequencies above that limit it is due mainly to differences of intensity. "Artificial" localization takes place, when we are listening to two sound sources placed at certain angles to the left and right in front of us and transmitting the same sound signal. Then we have the impression that the sound is coming from a source placed somewhere between the two real sources, the position of the "artificial" source being dependent upon differences of intensity and time of arrival of the two sound signals. We are able to change this apparant position of the "artificial" source by changing the intensity difference or the time difference or both of them, as has been shown by K. DE BOER [2]. It is however impractical to operate with the time difference, as it depends too much on the position of the listener. This means that we cannot in practical cases count on having a true stereophonic impression of the low frequencies, since the localization occurs by time differences. We must therefore operate with differences of intensity and try to find out the variation of the angle of the "artificial" source with this difference. DE BOER found experimentally this variation for a symmetrical position of the observer, the angle between the two loudspeakers being 50° approximately. At "Den danske Statsradiofoni" (The Danish Broadcasting Co.) we repeated and extended

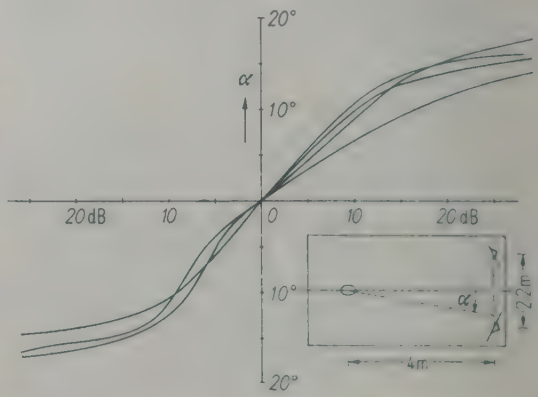


Fig. 1. Variation of the angle of the "artificial" source with the level difference in dB of the two loudspeakers. Symmetrical position of the observer. Angle between the loudspeakers 30°.

these experiments. For the symmetrical positions (angle between loudspeakers 30° or 60°) we found (Fig. 1) within a certain range proportionality between the level difference in dB and the angle of the artificial source although there were considerable variations in the curve for different observers. The slope of the curve was about 1 dB per degree when the angle between the loudspeakers was 30° and about 0.5 dB per degree when the angle between the loudspeakers was 60°. At an unsymmetrical position (with an angle between the loudspeakers of 30°) the linear part of the curve had a slope of 0.5 dB per degree (Fig. 2), whereas with an angle

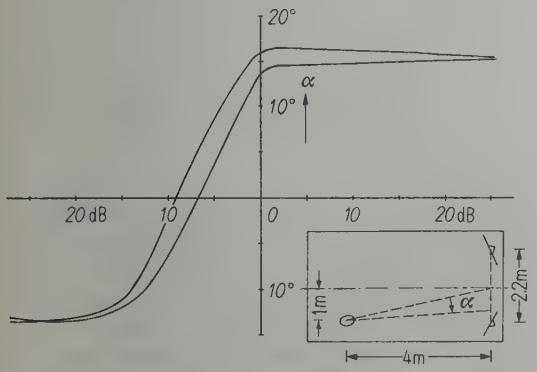


Fig. 2. As Fig. 1 but with unsymmetrical position of the observer, again 30°.

between the loudspeakers of 48° (Fig. 3) the results for different observers were very much different but the slope in dB per degree in any case still lower.

A tentative conclusion of these experiments was that for positions not too far from the axis of symmetry and not too close to, nor too far from the loudspeakers, a variation of 0.5 dB per degree would be a suitable mean value. It must be stressed,

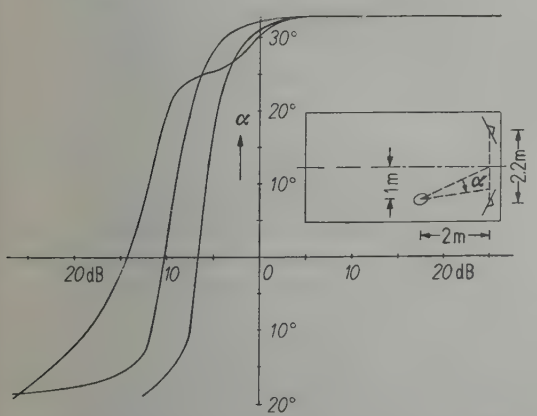


Fig. 3. As Fig. 2, but with an angle between the loudspeakers of 48° instead of 30°.

however, that the distortions of the stereophonic impression are rather large at the unsymmetrical positions, as is easily seen from the curves. These distortions must be neutralized by developing suitable directional properties of the loudspeakers. To obtain the desired variation in intensity level of 0.5 dB per degree we looked for microphones having directional characteristics which within the angle used for the stereophonic reproduction were logarithmic spirals. Outside this angle the sensitivity ought to be zero. This meant that we had to use highly directional microphones. When the system was first tried some years ago, we used unidirectional microphones as described in the literature [3]. They are ordinary moving coil microphones provided with a bundle of thin tubes of different length placed before the microphone. Fig. 4 shows two directional characteristics for such microphones at frequencies 1400 and 2000 c/s. The dotted curve in Fig. 4 is the ideal spiral having an increase in sensitivity of 0.25 dB per degree.

The experience gained with this system shows that we can obtain rather satisfactory stereophonic impressions both from orchestral music and from speech and sound effects, although the microphones correspond only roughly to the ideal ones. It is important that the recording studio as well as the listening room have low reverberation times, say 0.5 s for studios of 500 to 1000 m³ volume. It was observed, as might be expected, that sound origi-

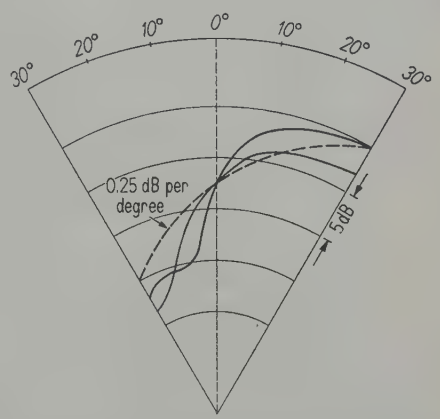


Fig. 4. Directional characteristics of "tube microphones" at frequencies 1400 and 2000 c/s (drawn in full) compared to the ideal characteristic for stereophonic transmission (dotted).

nating close to the axis of symmetry when reproduced gives a lower level than sound coming from the sides. To avoid this one must either reduce the angle between the microphones, thus reducing the area from which sound may be recorded, or one must place a third microphone (also highly direc-

tional) between the two others, connected through amplifiers to both loudspeaker channels. The lower limit for the directional efficiency of the microphones applied was 300 c/s, and it was observed that the reverberant sound below this limit was somewhat disturbing in the reproduced sound picture.

Recent practice with this system has shown that other types of microphones are applicable, e.g., condenser microphones placed in the foci of two parabolic reflectors or cardioid microphones of the modern type.

Speaking of further developments it seems appropriate to use either microphones which are directional also at the lowest frequencies to avoid the reverberant sound, or studios with very low reverberation times. Moreover the frequency band should be divided at 800 c/s and the frequencies below this limit should be transmitted to a single loudspeaker placed in the middle. Certain features of this system remind one of the system proposed by WARNCKE [4]. He proposed an array of highly directional microphones dividing the recording studio into narrow sectors. The microphones are connected to two loudspeaker channels via attenuation networks, the attenuation for a certain microphone depending upon its position in such a manner that a microphone on the left transmits most of its output to the left loudspeaker and vice versa.

It seems, however, as if the system here mentioned is somewhat simpler both in arrangement and in practical use.

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Discussion

RODRIGUES DE MIRANDA, J.: Having one loudspeaker only for frequencies up to 800 c/s it seems impossible to distinguish the direction of sounds of these frequencies?

LECTURER: Yes, but this does not matter so much, because the stereophonic impression in general is more attached to the higher frequencies.

MÖLLER, F. C. W.: I understand that the system explained requires a low reverberation time both for the transmitting and for the receiving room. Would not such low reverberation times have a bad influence on the musical impression, particularly for orchestral music?

LECTURER: The binaural effect seems to make reverberation less important. There is also the possibility of adding artificial reverberation.

BEGUN, S. J.: I wonder whether some additional time delay introduced between microphone and corresponding loudspeaker might not be helpful. Particularly in a hall which is very wide in relation to length intensity differences alone might not be satisfactory. In connection with the Cinerama system of sound reproduction, 6 channels were found to be a minimum to get some degree of realism.

LECTURER: The system described deliberately avoids time differences, since the time differences introduced by placing the microphones far from each other may disagree seriously with the time differences occurring by natural hearing. I do not feel convinced of the superiority of the multichannel systems.

THE RADIO SET AS AN INSTRUMENT FOR THE REPRODUCTION OF MUSIC

by J. RODRIGUES DE MIRANDA
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Summary. 1. Distortion should be decreased as the high frequency range increases. 2. Means are desirable for adjusting bass response, for cutting off and for gradual attenuation of treble. 3. The electro-acoustical engineer and the architect should cooperate closely. The cabinet must be rigid, the loudspeaker placed forward; its cloth chosen carefully. 4. The speaker should be chosen in accordance with the acoustical properties of the cabinet. The audio-frequency characteristic of the set should be carefully determined.

Sommaire. 1. Il faut réduire la distorsion lorsque l'étendue de la gamme haute fréquence augmente. 2. Il convient qu'il y ait des dispositifs appropriés pour régler la réponse des notes graves et pour couper ou affaiblir graduellement les notes aiguës. 3. Il doit y avoir une collaboration étroite entre l'électro-acousticien et l'architecte. La cabine doit être rigide, le haut-parleur étant monté à l'avant et muni d'un tissu soigneusement choisi. 4. Choisir le haut-parleur en fonction des caractéristiques de la cabine. La caractéristique des audio-fréquences de l'appareil doit être soigneusement déterminée.

Zusammenfassung. 1. Der Klirrfaktor soll niedriger sein, wenn das Diskantgebiet zunimmt. 2. Es ist erwunscht, daß Möglichkeiten vorhanden sind für die Tieftonreglung, das Abschneiden und das allmahliche Regeln der hohen Töne. 3. Der Elektroakustiker und der Konstrukteur sollen eng zusammenarbeiten. Das Gehäuse soll starkwandig sein, der Lautsprecher möglichst weit vorn montiert und die Bespannung der Lautsprecheröffnung sorgfältig ausgewählt werden. 4. Der Lautsprecher soll mit Rücksicht auf die akustischen Eigenschaften des Gehäuses gewählt werden. Die Frequenzkurve des Gerätes soll sorgfältig festgelegt werden.

This paper has been written from the viewpoint of an engineer who has to supervise the quality of reproduction of radio sets and who therefore has to deal with electrical as well as acoustical phenomena. However, an engineer will hardly ever be in a position to create the best possible amplifier and to design the best possible loudspeaking system as he has to take into consideration the financial and the aesthetical aspects too.

I will confine myself in this paper to better class radio sets and discuss some aspects of the design which lead to the manufacturing of apparatus from which real musical enjoyment may be expected.

I want to assume that we have at the input of the audio-frequency (a.f.) amplifier of a radio set an electric translation of the sound waves at the spot where we imagine the ear of the ideal listener to be.

This a.f. voltage has to be amplified and translated again into sound waves. One of the first principles the designing engineer has to adhere to is: to the sinusoidal voltages which together constitute the applied signal none other shall be added. This means that no non-linear elements should be present. As non-linearity cannot be avoided absolutely limits have to be given which should not be surpassed.

These limits can be given in two ways:

a) by the percentage of distortion,

b) by the percentage of intermodulation.

Usually the figures of merit of the amplifier and of the loudspeaker are given separately.

It will be clear that the distortion figures aimed at vary with the class of the radio set. When the distortion percentage method is used the percentage distortion should be given at a frequency of say 1000 c/s as well as at a low frequency e.g. 60 c/s, as the maximum output tends to drop at lower frequencies. The intermodulation figures can be judged by the percentage and by the value of the two frequencies.

The phase characteristic of the amplifier usually is unimportant as long as sharp rotations do not occur in the frequency range to be used.

Whilst it is obvious that any addition of frequencies may result in unwanted effects, the question of linear distortion, i.e. unequal amplification of the various frequencies is less clear. This linear distortion tends to alter the "colour" of the reproduction. Here we meet a typical and modern problem: is the quality of real music the quality that is liked best? It is generally known that about 90% of the tone controls of radio sets and gramophones remain in the position: treble cut-off. Is this so because the high tones are not reproduced faithfully or are they not appreciated in general?

A fact is that distortion at the high frequency side of the musical spectrum is very disagreeable or to be more exact — the components of the distortion products which fall in this region are very annoying. According to my experience this is the main reason why so often the high tones are cut off.

If we aim at high fidelity we simply have to strive for less distortion. In this connection it is necessary to point out that the harmonics of higher order are by far the more dangerous.

Let us again assume the most ideal conditions, to wit, that no distortion is present in the amplifier and that the acoustical characteristic of the loudspeaker with its enclosure is straight as a function of the frequency. Even in this case — which is only a hypothetical one — we know that for lower frequencies we often need an increased output. In normal radio cabinets the cross-over frequency varies from 100 to 300 c/s. It is possible and usual to correct the amplifier for this drop to a certain extent, but below the resonance frequency of the loudspeaker not much can be done.

Returning now to the treble end of the spectrum, other criteria are present. In the case of AM reception, atmospherics, man-made statics, interference whistles (at 9 kc/s) frequently necessitate a sharp cut-off, often combined with a 9 kc/s filter. The older of gramophone records asks for means for decreasing needle scratch. Usually only FM reception (at least in Europe), good long playing records and tape recorders justify the use of the full frequency spectrum.

Thus we find the desirability for sharp cut-off of treble, if possible with an adjustable cut-off frequency which in AM receivers might partially be accomplished in the i.f. part. In better class radio sets a bandwidth control therefore is opportune. As far as record playing equipment is concerned it is possible to give the pickup a sharp cut-off characteristic and/or to provide a suitable filter in the amplifier.

Besides this treble control it is advantageous to have in a radio set or gramophone means for adjusting the high frequency response curve. Such a control should provide an adjustable gradual dropping of the output towards the higher frequencies. This leads us to the conclusion that in order to fulfil all needs, two kinds of treble control are necessary.

I would like to offer the following prescription: give the user the possibility to utilize the whole audio spectrum as far as the basic design permits, but give him also the means to adapt the frequency characteristic to his needs and to his taste.

The design of the audio part of radio sets and gramophones in such a way that it fulfils the above mentioned requirements is more or less standard technique. The main task for the designing engineer

is: to compose commercially justified circuits and to obtain the requirements at a minimum cost price. The very low hum and noise levels asked for make his task even more difficult.

We turn now to the electro-acoustical transducing system, the loudspeaker with its cabinet. We can distinguish the table model, the console type and installations with a separately built loudspeaker enclosure.

The acoustical reproduction quality obtainable depends on the dimensions and the shape of the cabinet and on the loudspeaker characteristic. Usually there are general instructions regarding type of cabinet and its dimensions. It belongs to the task of the electro-acoustical engineer to find the best compromise for a certain amount of money, to find the most suitable type of loudspeaker and to adapt the amplifier to give the best results. It is also part of the work of this engineer to maintain the closest possible relation with the cabinet designing department in order to avoid unwanted constructions or dimensions.

He wants a rigid cabinet of heavy material. This is point one. Constructional considerations often ask for a baffle, fastened to the inner surface of the front of the cabinet. As a result the loudspeaker has to be mounted quite deep behind the outer surface of the cabinet. This influences the acoustical characteristic quite considerably as may be seen from the following curves.

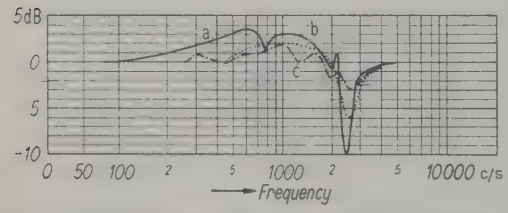


Fig. 1. Influence of baffle depth in front of the loudspeaker.

In Fig. 1 curves a, b and c illustrate respectively the behaviour of a baffle of 25, 16 and 10 mm thickness. As might be expected the dip always occurs at the same frequency, the depth depending on the thickness of the panel.

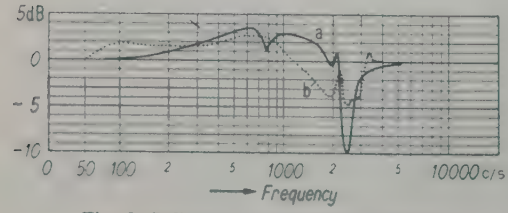


Fig. 2. Influence of microphone position.

In Fig. 2 the influence of the distance of the microphone on the measurements is shown. Curve a is identical with curve 1a, the distance being 50 cm, curve b gives the results for a distance of 10 cm. Increasing the distance to 100 cm gave no noticeable difference with curve 2a. Fig. 3 illustrates the influence of the size of the loudspeaker opening. Curve a again is the same as 1a being determined for a 8" loudspeaker, curve b is for a 6" speaker on a baffle of the same thickness.

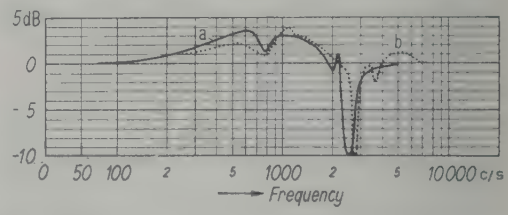


Fig. 3. Influence of the size of the loudspeaker opening.

It is obvious that as a rule the effect is unwanted especially as the dip occurs in the most vulnerable part of the frequency characteristic. For the reproduction of sufficient bass it is of paramount importance that as little acoustical leaking as possible is present. The resonance frequency of the speaker should lie about one octave below the cross-over point. A point which gives the electro-acoustical engineer much headache is the curious phenomenon that the architect, inspired by the customer-to-be, is inclined to spirit away the loudspeaker itself. Then again the loudspeaker opening has to be covered with a rich looking cloth. And this cloth can very well be the cause of poor quality reproduction. The influence of 3 types of loudspeaker cloths is shown in Fig. 4. The distance of the points of each curve from the zero-line is the attenuation in dB at the

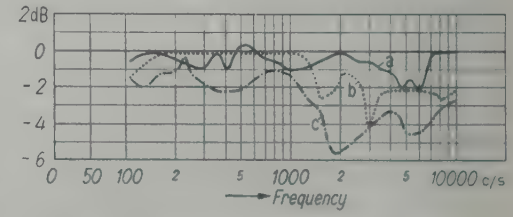


Fig. 4. Attenuation by loudspeaker cloths.

various frequencies. No comment is necessary. Now it is really difficult to design a cloth that is aesthetically adequate, that presents no difficulties in handling and glueing, that shows no unduly sagging or stretching under different atmospheric conditions and that is satisfactory from an acoustical point of view. The closeness of the weaving, the weight per unit of area, the stress at the process of glueing are important factors in the acoustical properties.

The bigger sets have an output of, say, 10 watts so that we have to decide between the use of two loudspeakers of smaller power-handling capacity or of one speaker which is capable of handling the 10 watts itself. The difficulty of course is that the heavier type speakers with low resonance frequency usually are not too good in the higher frequencies, whilst the use of two speakers gives rise to bundling of the higher frequencies.

The surface area of table type radio sets usually will not allow the use of two speakers of reasonable size. Fortunately with modern techniques the possibility exists of constructing a loudspeaker with a splendid frequency characteristic. As an example a characteristic of such a speaker is shown in Fig. 5,

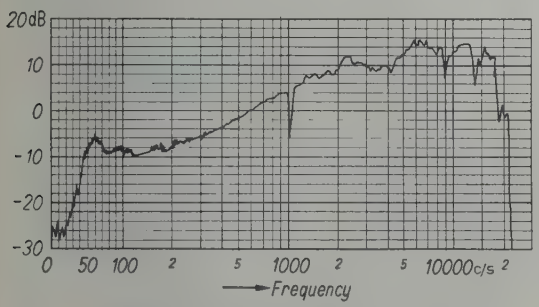


Fig. 5. Acoustical curve of a wide-range loudspeaker.

the resonance frequency being apparently 50 c/s, whilst the treble reproduction is extremely good up to 18000 c/s. The power-handling capacity is 10 W. The sound radiation diagram is amazingly good.

When the available space permits we might think of using an acoustical box or a bass reflex cabinet as an integral part of a floor model (console) radio set or radiogramophone. When the loudspeaker housing is a separate unit we have such a variety of enclosures at our disposal that it is an impractical object for discussion.

Nearly all systems have their advantages and disadvantages and it is often a matter of personal taste which leads to the decision. The only thing I want to mention in this regard is that technical information should be complete and reliable. Phrases as: range from 50 ··· 10 000 c/s are of no use at all but belong to bad advertising policy.

Discussion

Becking, A. G. Th.: Can the frequency of the dip, caused by mounting the speaker behind the baffle, be calculated? There seems to be a close similarity to body baffle dips in hearing aids.

Lecturer: The dip is probably due to interference between direct sound waves and sound waves reflected by the cylindrical wall of the baffle hole. Calculations confirm this theory to a close approximation.

Cramwinckel, A.: The lecturer aims at producing musical enjoyment but the question asked is: Is it generally correct to state that a listener is confronted with *real music* when he is listening to reproduced music either by electronic or by mechanical means, or will it always be a substitute (quite apart from considerations of more or less faithful reproduction)?

Lecturer: No publications known, but it is my opinion that the listener in fact is confronted with real music. As a comparison one might say that looking at a picture of a beautiful girl may give as much (or sometimes more) enjoyment as looking at the girl herself.

DIE CHARAKTERISTISCHEN EIGENSCHAFTEN MAGNETISCHER SCHALLAUFZEICHNUNGEN IN ABHÄNGIGKEIT VON DEN BETRIEBSBEDINGUNGEN UND DEN TONTRÄGEREIGENSCHAFTEN

von H. Schiesser

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Zusammenfassung. Die Abhängigkeit der Qualitätskenngrößen einer magnetischen Schallaufzeichnung von den Tonträgereigenschaften und den Betriebsbedingungen der Anlage wird gezeigt. Die physikalischen Tonträgereigenschaften bestimmen den Verlauf der statischen Remanenzkennlinie, die Betriebsbedingungen ihre durch die zyklische Magnetisierung beim HF-Verfahren bedingte Abänderung zur dynamischen Kennlinie. Der grundsätzliche Einfluß der Kennlinienform auf die Größen Empfindlichkeit, Pegel, Frequenzgang, nichtlineare Verzerrungen, Störgeräusch, Kopiereffekt und Löschbarkeit wird gezeigt und die Abhängigkeit dieser Größen von den physikalischen Tonträgereigenschaften wird experimentell ermittelt.

Summary. The relation between the quality factors of a magnetic sound record and the properties of the sound carrier and the conditions of operation is exemplified. The fundamental influence of the form of the characteristic on sensitivity, level, frequency response, non-linear distortion, background noise, printing-effect and erasure is pointed out and their relation to the physical properties of the sound carrier determined experimentally.

Sommaire. Les paramètres définissant la qualité d'un enregistrement magnétique du son dépendent du ruban et des conditions d'emploi de l'appareil. Les caractéristiques physiques du ruban

déterminent l'allure de la caractéristique statique de rémanance, tandis que les conditions d'emploi déterminent la caractéristique dynamique due à l'aimantation périodique. On a déterminé expérimentalement l'influence fondamentale de la forme de la caractéristique sur les grandeurs suivantes: sensibilité, niveau, réponse en fréquence, distorsions non linéaires, bruit, effet "echo", possibilité d'effacement, et sur leur dépendance vis à vis des caractéristiques physiques du ruban.

Über die grundsätzliche Arbeitsweise des Magnettonverfahrens besteht nach zahlreichen Untersuchungen der Einzelprobleme weitgehend Klarheit, dennoch sind wegen der Vielzahl der Parameter, insbesondere beim Aufsprechvorgang, kaum gesetzmäßige Beziehungen zwischen den physikalischen Eigenschaften der verwendeten Tonträger einerseits und den auf Magnettonanlagen erhaltenen elektroakustischen Kennwerten andererseits abzuleiten. Aus diesem Grunde wurden diese Zusammenhänge experimentell mit einer Standardanlage ermittelt und die möglichen Einflußfaktoren untersucht.

1. Physikalische Tonträgereigenschaften

Den Verlauf der Remanenzkurve B_r/H, die als statische Arbeitskennlinie anzusehen ist, bestimmen in erster Linie Koerzitivkraft und Sättigungsremanenz sowie die Anfangspermeabilität des Materials, Eigenschaften, die zweckmäßig an Bündeln in wechselstromdurchflossenen Solenoiden ermittelt werden. Untersucht wurden Materialien mit Koerzitivkräften bis 300 AW/cm und Remanenzen bis 800 G *).

2. Betriebskenngrößen der Magnettonanlagen

Die Arbeitsweisen der verschiedenen Magnettonanlagen sind sich in wesentlichen Punkten ähnlich: Verwendung einer Standardreihe von Tonträgergeschwindigkeiten, Anwendung des HF-Vormagnetisierungsverfahrens, Benutzung einseitig am Tonträger anliegender Ringköpfe von 10 ··· 20 μm Spaltbreite. Unterschiedlich sind Intensität und Frequenzgang der Vormagnetisierung sowie die schwer erfaßbare Form des Aufsprechspaltfeldes. Da hiervon Empfindlichkeit, Aussteuerbarkeit, Frequenzgang, Vormagnetisierungsstrombedarf, nichtlineare Verzerrungen, Störgeräusch, Kopiereffekt und Löschbarkeit weitgehend beeinflußt werden, wird für Tonträgerbeurteilungen eine Aufsprechanordnung vorgeschlagen, die aus einem reproduzierbaren Doppelleitersystem [1] definierten Feldverlaufs besteht. Für den Wiedergabekanal gestatten die vom CCIR aufgestellten Empfehlungen [2] die Erfassung der Oberflächeninduktion des Bandes mittels unterschiedlicher Abtasteinrichtungen.

3. Arbeitskennlinie

Die Anfangssteilheit der Remanenzkurve ist ein Maß für die Empfindlichkeit gegenüber einwirkenden äußeren Störfeldern. Bei zyklischer Vormagnetisierung wird der Aufsprechvorgang durch den Verlauf der dynamischen Kennlinie beschrieben, die sich von ihr durch geringere Krümmung im Ursprung und höhere Steilheit im mittleren Teil und Übergang in den Sättigungsbereich bei kleineren Feldstärken unterscheidet. Diese Abweichungen sind von der Intensität und der Frequenz der Vormagnetisierung und von den Signalfrequenzen, bzw. deren Wellenlängen, abhängig; für jeden Betriebsfall ergibt sich also eine andere dynamische Kennlinie. Sie ist ferner abhängig vom Feldstärkegradienten in Richtung senkrecht zum Tonträgerlauf und kann daher vergleichbar nur im homogenen Feld oder betriebsnäher in der erwähnten absoluten Aufsprechanordnung bestimmt werden. Anschaulich läßt sich ihre Entstehung in der zur Aufnahme der Hystereseschleifen benutzten Anordnung demonstrieren, wenn nach Vorschlag von SCHMIDBAUER zusätzlich ein Signalstrom entsprechenden zeitlichen Verlaufs zugeführt wird, so daß schrumpfende unsymmetrische Schleifen erhalten werden.

4. Empfindlichkeit

Der Empfindlichkeitsverlauf ist innerhalb des RAYLEIGH-Bereiches durch die Steilheit der Remanenzkurve gegeben; die festgestellten systematischen Abweichungen hiervon lassen sich auf das dem Spaltfeld überlagerte Feld des bereits magnetisierten Bandes zurückführen. Der Maximalwert der Empfindlichkeit ist für große Wellenlängen etwa proportional der Remanenz und der Schichtdicke (Abb. 1). Die hierfür erforderliche Vormagnetisierung steigt mit Koerzitivkraft und Schichtdicke (Abb. 2). Bei hoher Vormagnetisierung werden in der unmit-

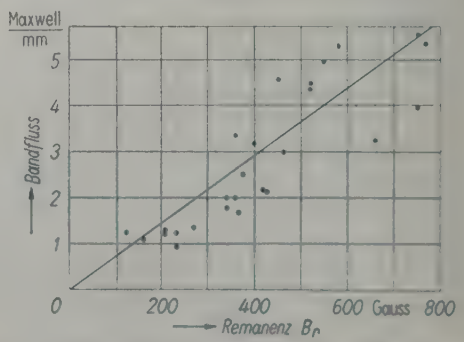


Abb. 1. Abhängigkeit des spezifischen Bandflusses von der Remanenz; $\lambda = 762$ μm; Arbeitspunkt: maximale Empfindlichkeit; $f_{HF} = 80$ kHz. 10^8 Maxwell = 1 Vs.

*) 10^4 Gauß = 1 V s/m².

telbar am Kopf anliegenden Zone die optimalen Aufzeichnungsbedingungen überschritten; die Aufzeichnung findet daher in Laufrichtung verschoben an einer Stelle mit flacherem Feldverlauf statt. Die

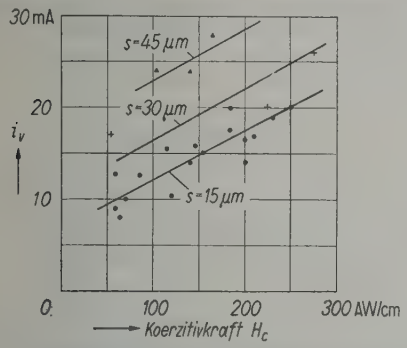


Abb. 2. Abhängigkeit des Vormagnetisierungsstromes für maximale Empfindlichkeit von Koerzitivkraft und Schichtdicke; $f_{NF} = 1$ kHz, $f_{HF} = 80$ kHz, $v = 76{,}2$ cm/s.

damit verbundene Benachteiligung der kleinen Wellenlängen ergibt für diese eine Verschiebung der Vormagnetisierung für optimale Empfindlichkeit zu kleineren Werten. Sie ist der Schichtdicke proportional.

5. Frequenzgang

Der Frequenzgang der Aufsprechentmagnetisierung ergibt sich aus dem Unterschied der dynamischen Arbeitskennlinien für die zeitlich entsprechend verlaufenden Zyklen im Aufsprechfeld. Abweichend vom zu erwartenden exponentiellen Verlauf tritt bei bestimmten Vormagnetisierungswerten ein Einbruch im Frequenzgang auf, der ebenfalls auf die Rückwirkung des bereits magnetisierten Bandes auf den Feldverlauf zurückzuführen ist. Der Exponent des Abfalles, mit der absoluten Aufsprechanordnung bestimmt, stellt ein vergleichbares Maß für den Frequenzgang eines Tonträgers bei gegebenen Betriebsverhältnissen dar. Die Selbstentmagnetisierung des Tonträgers, die durch Rückscherung am Wiedergabekopf zum großen Teil rückgängig gemacht wird, ist ein weiteres Bestimmungselement des Frequenzganges. Der Empfindlichkeitsrückgang eines Bandes bei hohen Frequenzen, bezogen auf die spezifische Empfindlichkeit bei niederen Frequenzen, ist im wesentlichen von der Koerzitivkraft und damit von der Anfangspermeabilität abhängig.

6. Nichtlineare Verzerrungen

Die kubischen, allein interessierenden Verzerrungen sind aus der ersten und dritten Ableitung der Remanenzkurve zu ermitteln, wobei eine Integration über alle an der Aufzeichnung teilnehmenden Schichten erforderlich ist. Abweichungen hiervon treten durch stärkere Kennlinienkrümmung bei größerer Zyklenzahl auf. Der Wendepunkt der Kennlinie führt zu einem Minimum der Verzerrungen in diesem Arbeitspunkt. Unter Betriebsbedingungen ist der für gegebene Verklirrung erzielbare Bandfluß eine Funktion der Remanenz und steigt bei großen Wellenlängen etwa mit der Quadratwurzel aus der Schichtdicke (Abb. 3).

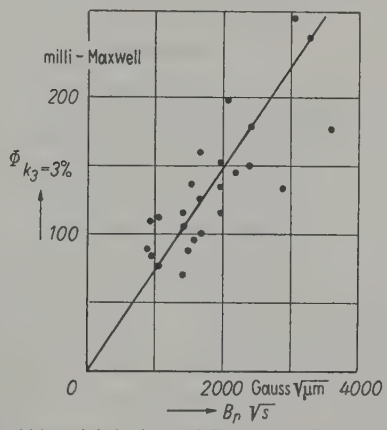


Abb. 3. Abhängigkeit des erzielbaren Pegels von Remanenz und Schichtdicke; $\lambda = 762$ µm; Arbeitspunkt: maximale Empfindlichkeit; $f_{HF} = 80$ kHz.

7. Störgeräusch

Neben einem durch die gesättigten Elementarmagnete gegebenen Ruherauschen tritt ein durch Amplituden- oder Frequenzmodulation des aufgezeichneten Signals oder der aufgezeichneten Vormagnetisierungsfrequenz verursachtes Störgeräusch auf. Im Falle der Amplitudenmodulation ist es durch die Inhomogenitäten des magnetischen Belages und des Kontaktes zwischen Band und Kopf, im Falle der Frequenzmodulation durch Bandlängsschwingungen verursacht. Wenn Ände-

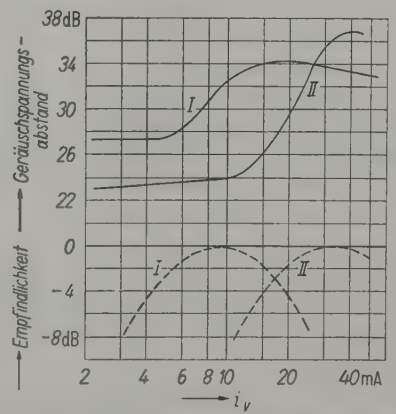
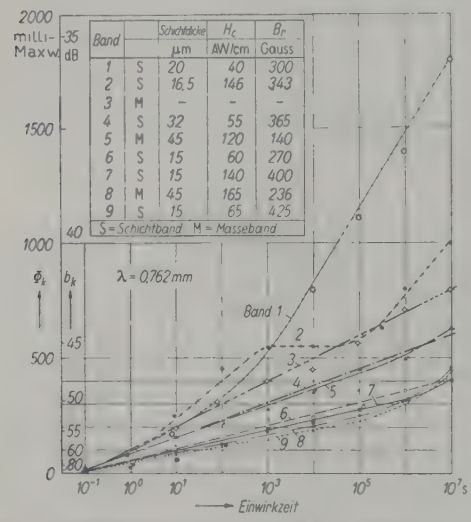


Abb. 4. Störgeräusch in Abhängigkeit vom Magnetisierungsstrom; $v = 76{,}2$ cm/s.

rungen des magnetischen Leitwertes im Aufsprechspaltfeld zu einer Schwankung des Aufsprechflusses führen, tritt eine Verstärkung des Störgeräusches ein. Oberhalb des Arbeitspunktes maximaler Empfindlichkeit vermindert sich der Einfluß des durch Kontakt- und Oberflächeninhomogenitäten verursachten Rauschens und der durch innere Bandinhomogenitäten verursachte Anteil bleibt übrig (Abb. 4.).

8. Kopiereffekt

Der von der Anfangssteilheit der Remanenzkurve abhängige spontane Anteil ist bereits bei nur Sekunden dauernder Einwirkung zwischen den Nachbarwindungen gegenüber dem zeitabhängigen Anteil vernachlässigbar. Dieser zeigt bei einigen Bandtypen einen streng logarithmischen Verlauf über acht Zehnerpotenzen der Einwirkzeit, bei anderen reproduzierbare Sprünge oder stärkeren Anstieg bei längeren Einwirkzeiten (Abb. 5). Der Faktor des Anstiegs



Band		Schichtdicke µm	H_c AW/cm	B_r Gauss
1	S	20	40	300
2	S	16,5	146	343
3	M	–	–	–
4	S	32	55	365
5	M	45	120	140
6	S	15	60	270
7	S	15	140	400
8	M	45	165	236
9	S	15	65	4,25

S = Schichtband M = Masseband

Abb. 5. Bandfluß Φ_k der Kopie und Kopierdämpfung b_k in Abhängigkeit von der Einwirkzeit.

läßt keine Zuordnung zu den magnetischen Bandeigenschaften erkennen; er wird auf intermolekulare Vorgänge, für die die Aktivierungstemperatur ein Maß ist, zurückgeführt. Ein überraschender Einfluß mechanischer Bandbeanspruchungen auf den Kopiereffekt wurde festgestellt, als dessen Ursache Magnetostriktion vermutet wird. Diese Tatsache zeigt neue Möglichkeiten sowohl zur Herstellung von Kontaktkopien als auch zur Verbesserung der Kopierdämpfung ohne Verwendung eines die Güte des Originals bzw. der Nutzmodulation beeinflussenden Hilfsfeldes.

9. Löschbarkeit

Die grundsätzlichen Zusammenhänge zwischen Löschbarkeit und Koerzitivkraft zeigen starke Anomalien, auch in bezug auf den Einfluß der Stabilität der zu löschenden Aufzeichnungen und die Verbesserung der Löschwirkung bei Mehrfachlöschung sowie die Wirkung der zwischen den Einzellöschungen liegenden Zeiträume. Auch der Einfluß der Löschfeldrichtung ist für einzelne Bandtypen stark unterschiedlich und läßt noch keine Gesetzmäßigkeiten erkennen. Es wird vermutet, daß Nachwirkungseffekte die Ursache dieser Erscheinungen sind.

Schrifttum

[1] Schmidbauer, O., Der absolute Bezugssprechkopf. Mitteilungen des Rundfunk-Technischen Instituts Nr. 12/1952.

[2] CCIR, 1952, Studiengruppe X, Dokument 24.

Diskussion

Begun, S. J.: a) From Mr. Schiesser's remarks it becomes obvious that the forces acting on the magnetic particles are small indeed. One only has to consider the fact that the particles are imbedded in an elastic medium which will absorb most of the deformation stresses, and thus further reduce the forces on the particles. I believe more work should be done in this field, to learn more about the mechanism of this phenomenon.

b) What is the order of magnitude of the mechanical forces which are required to transfer a signal from a mother to a daughter tape?

Vortragender: a) Mechanical forces acting on the magnetic particles are small indeed. The effect seems to be due to magnetostriction. Any changes in permeability whilst superimposing magnetic forces could not be found. Experiments on this problem are being continued.

b) The effect was observed by pressing the tape between capstan and rubber pressure roll. Stronger effects could be found when leading the tapes over a sharp edge or around a number of small rolls. Mechanical forces are small enough to exclude any damage of the tape.

EINIGE SCHALLPLATTENAUFNAHMEPROBLEME

von F. SCHLEGEL
Fonofilm Industri A/S, Kopenhagen

Zusammenfassung. Einige Schallplattenaufnahmeprobleme werden erörtert unter besonderer Berücksichtigung der nur durch Gegenkopplung gedämpften Schreiber. Einzelheiten über einen in der Literatur der Hauptsprachen nicht beschriebenen Schreiber werden angegeben.

Summary. Certain problems in sound disc recording are discussed with special reference to cutting heads damped by countercoupling only. Data are given on a cutting head not described in one of the principal languages.

Sommaire. On discute quelques problèmes d'enregistrement sur disques, en considérant particulièrement le cas d'un enregistreur amorti seulement par réaction. On donne quelques détails relatifs à un enregistreur qui n'est pas décrit dans la littérature des principales langues.

Die Schallplattenaufnahme ist in der neueren Zeit wesentlich verbessert worden durch die Einführung gegengekoppelter Schreiber. Diese bieten ganz besondere Vorteile, wenn die gesamte Dämpfung des schwingenden Systems rein elektrisch gemacht wird, aber auch verstärkertechnisch die größten Schwierigkeiten. Der erste solche Schreiber wurde wohl von den Herren VIETH, WIEBUSCH und YENZER im Herbst 1938 beschrieben [1]. Dieser Schreiber ist ein „hill and dale" Schreiber, und wohl etwas leichter zu konstruieren als ein Schreiber für das Transversalverfahren, da es mechanisch leichter ist, einem hin- und herschwingenden System mit Hilfe von Parallelführungen nur einen Freiheitsgrad zu geben als einem System, welches sich um eine Achse dreht.

Es wurde in Dänemark während des Krieges versucht einen solchen Schreiber zu konstruieren, und im September 1949 veröffentlichte auch Herr YENZER, einer der drei obengenannten Herren, einen „Lateral feedback disc recorder" [2]. Da außer den Patentschriften über die dänische Konstruktion in großen Sprachen nichts veröffentlich worden ist, sei es mir gestattet, über diesen mittlerweile gut eingeführten Schreiber einige Einzelheiten zu berichten.

Wenn die Dämpfung eines Schreibers rein elektrisch gemacht wird, kann man erreichen, daß der Ausschlag praktisch nicht mehr vom Schneidewiderstand abhängt. Man kann den wirklichen Ausschlag jederzeit an der Gegenkopplungsspule registrieren und feststellen, daß der Amplitudenunterschied beim Schneiden von Lackplatten in Innenrillen und beim Schwingen in Luft kleiner als 1 dB ist. Erhält man einen schlechteren Frequenzgang auf der Platte, als er nach Beobachten der gegengekoppelten Spannung zu erwarten wäre, ist daraus zu schließen, daß der Stichel die Rillenwand nur elastisch deformiert hat. Bei hohen Frequenzen ist dies durchaus möglich, da ja die größten Amplituden, die man aufzeichnen wird, nur einige Mikron Ausschlag haben. Mit Hilfe solcher Schreiber lassen sich also leicht Versuche über die Schneidefähigkeit der Stichel durchführen. Schliff, Facetten und bei geheizten Sticheln die notwendige Temperatur können untersucht werden.

Die Wirkungsweise eines gegengekoppelten Schreibers kann leicht verständlich folgendermaßen erklärt werden. Ein schwingendes System mit nur einem Freiheitsgrad bewegt sich nach der Gleichung $m\ddot{x} + r\dot{x} + dx = p(t)$.

Ein solches System wird gedämpft durch die $p(t)$ entgegengesetzte Kraft $-r\dot{x}$, die also in Phase mit der Geschwindigkeitsamplitude und dieser proportional ist. In der Treibspule des Systems können wir uns zwei entgegengesetzte Ströme vorstellen; einen, der der treibende ist und die Kraft $p(t)$ erzeugt gemäß dem zugeführten Sprechstrom, und einen bremsenden, erzeugt von der Spannung über die Gegenkopplungsspule $-r\dot{x}$. Die Geschwindigkeit des Systems ist bekanntlich nur im Resonanzpunkt mit der treibenden Kraft in Phase. Bei großer Dämpfung ist die Geschwindigkeitsamplitude eines Systems bei gleicher Treibkraft fast frequenzunabhängig innerhalb eines breiten Bandes. Die Gegenkopplungsspannung ist also fast konstant, die Größe der Gegenkopplung selbst ist aber etwa Null an den Enden des Bandes und muß in der Mitte desselben einen Wert erreichen, der wesentlich von der unvermeidlichen Restdämpfung des Systemes abhängt und bei allen erwähnten Schreibern in der Gegend von 40 ··· 50 dB liegt. Diese Gegenkopplung ist daher nur von der Phasenlage der beiden Ströme durch die Treibspule oder von der Phasenlage zwischen $p(t)$ und $-r\dot{x}$ abhängig, die in richtigem Maße vom Schwingungszustand des Systems selbst geregelt wird. Der verwendete Verstärker muß innerhalb des Bandes absolut phasenrein sein, d.h. die Laufzeit aller Frequenzen durch diesen Verstärker muß Null sein. Schwingt das System anomal, wenn es z.B. mehr als einen Freiheitsgrad hat, so wird der Phasenzustand nicht mehr richtig sein. Schlimmstenfalls kann sich bei 180° Phasendrehung die

Gegenkopplung in volle Rückkopplung verwandeln, was katastrophale Folgen hat. Ein solches schwingendes System wird versuchen, sich bei großen Geschwindigkeitsamplituden um seinen Schwerpunkt zu drehen und damit seine elastische Lagerung zu deformieren. Wenn das System nur einen Freiheitsgrad haben soll, muß diese Aufhängung sehr stabil sein, was einen verhältnismäßig unempfindlichen Schreiber ergibt. Eine Aufhängung in Achslagern verbietet sich wegen der kleinen Amplituden. In deren Größenordnung wird ein Achslager immer Spiel haben, also gar keine Sicherheit für nur einen Freiheitsgrad bieten. YENZER beschreibt auch in seinem Artikel die Schwierigkeiten mit der Bewegung um den Schwerpunkt. Bei der Konstruktion des dänischen Schreibers beachtete man von vornherein dieses Problem und baute das System, wie aus der Abb. 1 hervorgeht, so, daß der Drehpunkt mit dem Schwerpunkt des Systemes zusammenfiel. Es gelang auf diese Art einen stabilen Schreiber großer Empfindlichkeit zu schaffen. Das Magnetsystem konnte so klein gehalten werden, daß der Schreiber nur 270 g wiegt. Auf Grund der Erkenntnis, daß es auch schwer sein könnte einen Saphirhalter für die höchsten aufzuzeichnenden Frequenzen völlig steif zu machen, wurde die Gegenkopplungsspule in Saphirnähe angebracht. Hat doch z.B. ein einseitig eingeklemmter Stab von 1 mm Durchmesser und 10 mm Länge aus Stahl oder Duraluminium nur eine Eigenfrequenz von 3800 Hz und ein solcher von 2 mm Durchmesser und 5 mm Länge eine solche von 30 kHz. Um die mechanische Stabilität des schwingenden Systems nicht zu verkleinern, wurde die Spule in einem Magnetfeld so angeordnet, wie es aus der Abb. 2 hervorgeht.

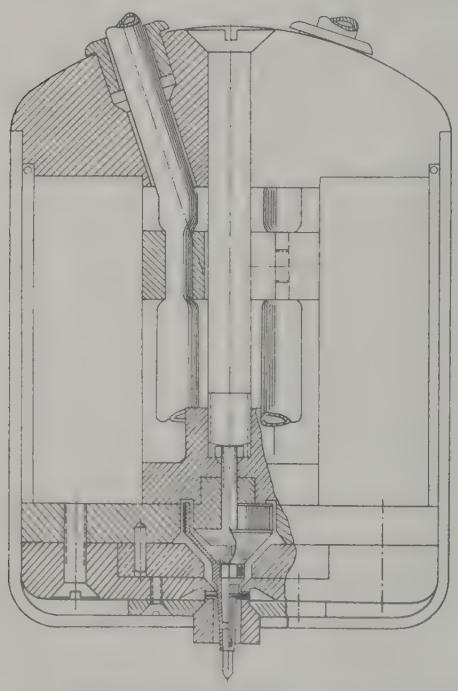


Abb. 1. Querschnitt eines elektrodynamischen Schallplattenschreibers.

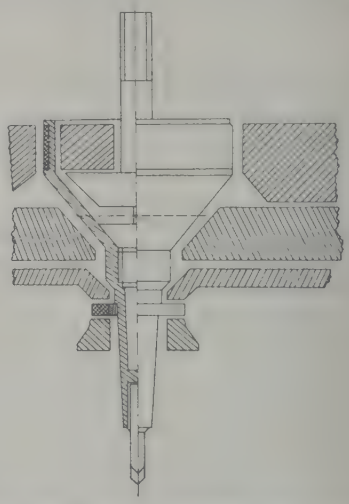


Abb. 2. Querschnitt des schwingenden Systems der Abb. 1.

Der Wirkungsgrad dieses Polsystems ist ausreichend, da man mit der aus anderen Gründen notwendigen Mindestanzahl von Verstärkerstufen auskommt.

Jedes frequenzbandbegrenzende Glied muß die Phase von $+ 90°$ auf $- 90°$ von Grenze zu Grenze drehen. Daher wird bereits bei nur einer Verstärkerstufe eine Phasendrehung von 180° und damit volle Rückkopplung möglich. Vermeiden läßt sich dies nur durch hinreichend große Dämpfung im ersten frequenzbegrenzenden Glied, ehe das zweite anfängt die Phase zu drehen, und durch besonderen Phasengang entsprechend NYQUISTS Theorem. Durch Ausnutzung der Steifigkeit des schwingenden Systems und phasenbeeinflussender Komponenten im Gegenkopplungskreis läßt sich eine große Dämpfung unmittelbar jenseits des hörbaren Frequenzgebietes erreichen, so daß eine genügende Stabilität der Verstärkerschaltung trotz des großen Gegenkopplungsgrades zu erlangen ist.

Doch wird durch die Gegenkopplung nicht nur der Vorteil einer großen Dämpfung erreicht, sondern auch der einer großen Reinheit der Aufzeichnung. Mit den genannten Schreibern lassen sich bei voller Aussteuerung und an den Grenzen des Bandes, wo die Gegenkopplung noch klein ist, Intermodulationen von einigen Promille erreichen, während die Intermodulation zwischen Frequenzen in der Bandmitte fast unmeßbar klein ist, in der Gegend von

0,1 ··· 0,2‰. Dieser Wert ist an der Gegenkopplung gemessen, während der Schreiber eine Platte schneidet. Ganz gewiß treten beim Abtasten der Platte Verzerrungen auf, die größer sind, aber sich doch, besonders bei Mikrorillen, in sehr bescheidenen Grenzen halten lassen. Die leichte Anbringung großer Schwungmassen in den Plattentellern ermöglicht es, die Geschwindigkeitsschwankungen sehr niedrig zu halten, Werte von 1‰ und weniger für Aufnahme und Wiedergabe zusammen sind durchaus zu ermöglichen. Daher stellt wohl heute die Schallplatte die besterreichbare Tonreproduktion dar, die wohl in den genannten Punkten sogar die Magnettonaufzeichnung übertreffen dürfte.

Durch Verwendung von Vinolyte und Polystyrol als Plattenmaterial und die Anwendung von geheizten Sticheln bei der Aufnahme kann das Nadelgeräusch moderner Aufnahmen sehr niedrig gehalten werden. Leichtes und billiges Produzieren der Platten, billige Abspielgeräte, Übersichtlichkeit einer Diskothek, sofortige Griffbereitschaft einer Aufzeichnung oder eines Teiles derselben sind die großen Vorteile der Schallplatte, die dieser wohl trotz aller Konkurrenz noch ein längeres Leben sichern werden.

Vom Standpunkt des Musikliebenden und auch des Technikers wäre zu wünschen, daß man mehr dazu überginge zur Pressung nur Vatermatrizen zu benutzen. Das Vervielfältigen einer Aufnahme, die ja fast immer auf Magnetophonband vorliegt, ist heute so gut möglich, daß man für jede Preßmatrize eine neue Lackaufnahme opfern könnte. Hierdurch würde man die genauest mögliche Preßmatrize erhalten, wenn schon Vervielfältigungen von Matrizen auf galvanischem Wege mit einer erstaunlichen Genauigkeit erfolgen können. Aber die kleinsten Amplituden höchster Frequenzen, die noch wiedergegeben werden können, liegen nun einmal in der Gegend von Lichtwellenlängen. Besonders beim Übergang zu gespritzten Platten, wo der Verschleiß der Matrizen bedeutend geringer wird, ließe sich dieser Gedanke vielleicht verwirklichen. Und die größte Bitte aller Musikliebenden, Platten ohne „Flutter". 1‰ zusätzliche Geschwindigkeitsänderungen werden gelegentlich schon hörbar. Soll dieser Betrag auf Innenrillen mit vielleicht 50 mm Radius noch nicht überschritten werden, so darf die Exzentrizität des Loches nur einige Hundertstel mm sein. In einem soeben erschienenen Standardvorschlag ist von einer etwa zehnfachen Toleranz die Rede. Wir Techniker müssen wohl umgehend eine Methode zur Erreichung dieses Zieles angeben, die so einfach ist, daß die Fabrikanten nicht davor zurückschrecken diese auch anzuwenden.

Schließlich noch ein kurzer Hinweis. Film, Band, vielleicht auch die Platte selbst kann vergänglich sein. Mit Draht aus Valdemar Poulsens Telegrafon hat man Erfahrung mit 50jähriger Lagerfähigkeit. Aber eine Preßmatrize, speziell wenn deren Oberfläche mit edlem Metall überzogen oder durch Aufdampfen geeigneter Substanzen wie Quarz oder Siliziummonoxyd gegen Korrosion geschützt wird, dürfte eine Lagerfähigkeit von Jahrhunderten, ja vielleicht Jahrtausenden haben und damit eine wirklich sichere Archivierung kulturell wertvoller Dinge ermöglichen.

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Diskussion

Skudrzyk, E.: Sind die Phasendrehungen im Bereich der Nadelresonanz und der tiefen Frequenzen für diesen Schreiber bekannt? Wie weit äußern sie sich in Laufzeiten und klanglichen Störungen, wie sie z.B. bei stark gegengekoppelten Radioapparaten in den falschen Bässen bekannt sind?

Vortragender: Wegen der hohen Resonanzfrequenz des Systems herrscht bei den tiefen Frequenzen voreilende Phase, die aber gebremst wird. Eine Korrektion der Phase im Vorverstärker wäre möglich. Der maximale Energieverbrauch ohne Gegenkopplung ist 1 ··· 2 Watt, es können aber Spitzen bis zu 15 Watt und außerordentlich kurzer Dauer (10 ··· 50 µs) auftreten, die auf Bremsung der angestoßenen Resonanz durch die Gegenkopplung zurückzuführen sind.

Rademakers, A.: Können Sie Zahlen angeben in bezug auf die Differenz zwischen Stichelamplitude und geschnittener Amplitude?

Vortragender: Keine bei tiefen Frequenzen; unter guten Bedingungen findet man bei 15 kHz höchstens 2 ··· 3 dB Abweichung gegen die Messungen mit einem Tonabnehmer Ortofon C mit nur 1 mg Spitzenmasse und den damit zu erwartenden theoretischen Wert und gegen das Meyerlicht, dessen Angabe aber wegen der auftretenden Interferenzstreifen auch ungenau ist.

Franz, W. weist auf die Schwierigkeiten bei der Messung der Tonhöhenschwankung und der Schütterspannung von sehr hochwertigen Schallplattenabspielmaschinen hin, weil diese Maschinen günstigere Daten aufweisen als die zur Zeit zur Verfügung stehenden Prüfplatten und Leerrillenplatten.

THE ESTABLISHMENT OF QUALITY STANDARDS BY SUBJECTIVE ASSESSMENT

by T. Somerville

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Summary. B.B.C. methods for subjective testing in the field of electro-acoustics are described. The B.B.C. endeavours to reproduce the original sound because only in this way is a reliable standard available. From considerable experience it has been found to be quite impossible to make reliable comparisons subjectively unless the acoustics of the originating studio are good. Loudspeaker testing is used as an example, although the methods are applicable to other types of electro-acoustic equipment.

Sommaire. On décrit les méthodes d'essais subjectifs employées en électro-acoustique par la B.B.C. La B.B.C. s'efforce de reproduire le son original, car c'est la seule façon d'obtenir un jugement sûr. D'après la très grande expérience acquise en la matière, il est tout à fait impossible de faire subjectivement des comparaisons sûres, si l'acoustique du studio d'essai n'est pas bonne. On a pris comme exemple l'essai d'un haut-parleur, mais les méthodes sont applicables à d'autres types d'appareil électro-acoustique.

Zusammenfassung. Es werden einige bei der B.B.C. benutzte subjektive Testmethoden beschrieben. Dabei ist man stets bestrebt, die Originalklänge wiederzugeben, weil man nur auf diese Weise eine zuverlässige Norm erhält. Auf Grund zahlreicher Erfahrungen hat sich gezeigt, daß man zuverlässige subjektive Vergleiche nur durchführen kann, wenn die Akustik des Aufnahmestudios gut ist. Als Beispiel wird die Lautsprecherprüfung beschrieben; jedoch sind die Methoden auch auf andere elektroakustische Geräte anwendbar.

The maintenance and improvement of quality standards in any electro-acoustic system, whether it be broadcasting or sound recording, requires constant supervision, a comprehensive measuring technique, and a reliable system of subjective assessment. There has been considerable development in measuring techniques, but there is so far no established system of subjective tests. Yet it is the subjective assessment which provides the final verdict on the capabilities of a system.

Listening tests of one type or another are always carried out, but there are pitfalls in subjective testing. It may therefore be of advantage to outline methods used in the B.B.C. as a guide to other workers.

No doubt it will be generally agreed that reproduction of sound by electro-acoustic means can never be an exact replica of the original. This is so for a number of reasons. First and most important, most recording and broadcasting systems are not stereophonic; it is not economic to employ more than one channel between the listener and the studio. There are also inevitable distortions of the signal in every part of the system, so that it is quite impossible to reproduce the original exactly.

As the original sound cannot be duplicated, two fundamental decisions have to be made. The first decision by most organisations is that every attempt should be made to reproduce the original as closely as possible, because only in this way can a standard, reasonably unaffected by matters of opinion, be obtained.

As the listening room acoustics also have a profound effect on the results, a decision is necessary on the standard listening conditions. It is often argued that listening rooms should be as "dead" as possible so that their acoustics cannot have an appreciable effect on the programme. In this connection it should not be forgotten that the polar diagram of the loudspeaker also affects the subjective reverberation time of the room. Some organisations make studio listening cubicles dead; they perhaps do not realise that monitoring of the programme may be misleading because the ordinary listener does not listen under very dead conditions. Experiments have shown that in Britain the reverberation time of the average living room is of the order of half a second, and for this reason most of the B.B.C.'s critical subjective tests are carried out in listening rooms having a reverberation time of this order. These rooms are furnished as far as possible in the manner of living rooms.

In carrying out subjective tests it is usual to proceed by a series of successive assessments, each part of the chain being dealt with in turn. Such an investigation becomes very elaborate and must be conducted with care if reliable results are to be obtained. It should be noted that it is quite possible for one element of the chain to compensate for faults in another, so that improvement in one part sometimes results in an unexpected deterioration in overall performance. The best way to illustrate some of the difficulties is to describe some experiments conducted by the B.B.C.

After the war it became necessary to select a new loudspeaker for monitoring B.B.C. programmes.

Manufacturers were therefore asked to submit samples of high-grade loudspeakers for tests. The observers taking part in the subjective assessments were not given the results of these tests until their opinions had been recorded.

The first subjective comparisons were made in a listening room on the B.B.C. programmes available. It was found that although the majority of the loudspeakers could be rejected as unsuitable, some appeared to be satisfactory only on certain types of programme. Furthermore, the order of preference was found to vary with the programme material. It was soon realised that a decision could not be made unless comparison with the original programme was possible, and so the next series of tests was carried out at a studio centre.

Here again the results were inconclusive for the same reason, and it was further noted that wide-range loudspeakers were normally disliked because of harsh quality. Careful listening tests conducted in the studio led to the conclusion that the acoustics were indifferent. The tonal quality was harsh, and in loud passages the weaker instruments became inaudible.

To discover whether or not the anomalies were due to poor acoustics the whole experiment was transferred to a good orchestral studio, whereupon it was quickly found that reliable comparisons could be made resulting in the rejection of all but three of the loudspeakers under test. The three remaining loudspeakers were a wide-range unit of American origin, and two single-cone units of an early type (A and B) both made by the same manufacturer. It is interesting that this was the first occasion on which a wide-range loudspeaker gave pleasing results.

At this juncture it was realised that the loudspeaker employed in the studio cubicle was type B. Lest this should be affecting the choice, type B was replaced by type A with quite surprising results. The microphone had to be moved much further from the orchestra and raised higher to find the best position, which then gave much superior results on the wide-range loudspeaker and caused the rejection of the single-cone unit type B. To complete the experiment the orchestra was "balanced" on the wide-range unit and gave satisfactory results on the single-cone unit type A.

The moral behind this experiment is that excellent acoustics are essential for such tests; furthermore the loudspeaker used can have profound effects on the result and should therefore be beyond reproach.

Profiting by this experience a further critical test for loudspeakers was evolved. Since the balance or the microphone position depends very much on the loudspeaker, it was decided in future always to allow the staff responsible for balancing to try new loudspeakers. When this is done the microphone position corresponding to each loudspeaker will vary and therefore only those loudspeakers which correspond to approximately the same microphone position can be used with safety. Any loudspeaker which makes the operator place the microphone in an unusual position, for example, very close to the orchestra, is suspect, because obviously corrections are being made for deficiencies in the loudspeaker.

Another important test for all electro-acoustic equipment is to use speech from a non-reverberant room or from outdoors. This becomes a very stringent test when the speaker's voice is known. It is surprising that in spite of its simplicity and value very few manufacturers make use of such a speech test from the open air. A skilled observer can pick out quite easily the regions in which colourations exist, but he will frequently find it difficult to convince other people.

An excellent demonstration can be organised by placing a speaker in the non-reverberant room and recording his voice, replaying it into the non-reverberant room on the loudspeaker under test, recording and replaying again and again. This operation, which is equivalent to dubbing in recording systems, accentuates the peculiarities and makes them very audible. All defects of the system are magnified by this process, but as the loudspeaker is the weakest link in the chain its faults predominate.

The loudspeaker tests described serve to illustrate the difficulties of subjective testing and particularly the need for a reliable source of programme. This is not generally realised by manufacturers, who frequently carry out tests on broadcast programmes or on gramophone records. They are then entirely in the hands of the broadcasting organisation or the recording company, which for example may have used a loudspeaker with peculiar properties in arriving at the balance. It is not an exaggeration to say that if such a loudspeaker is employed it is only possible to get correct results by employing a similar loudspeaker for listening.

Similar methods are adopted in the subjective testing of microphones. Here again the best method is to use speech and also to use the microphone on an orchestral concert in a good studio. It is sometimes more convenient to make comparisons at rehearsals. In this connection a word of caution is necessary, for orchestras frequently do not play well at rehearsals and some of the faults, for example poor intonation between instruments in the same section, can cause a roughness which may sometimes be mistaken for non-linear distortion.

Recording systems can also be tested by comparing the original material with the recorded and reproduced result. In this case dubbing is a very potent method of showing up deficiencies in the recording system. It is usual to find that after four dubbings the faults in even the best recording system have become noticeable.

Thanks are due to the Chief Engineer of the B.B.C. for permission to publish this paper.

Discussion

KLEIS, D.: Is it not a difficulty that by reiterating in a non-reverberant room allowance has to be made for the directional characteristics of the loudspeaker, which are smoothed out in the normal listening room?

LECTURER: There is no great difficulty. The colourations can be observed on the axis of the loudspeaker and the tests are made with the microphone in this position. It should also be remembered that, if very reverberant conditions are employed, the loudspeaker colourations may be completely obscured.

SMIT, H. J. A.: Was there any correlation between the preference in subjective tests and the measured curves of loudspeakers?

LECTURER: The correlation was good.

RODRIGUES DE MIRANDA, J.: At Philips a "living room" is always used for listening tests, the room being equipped with switch-over arrangements for chassis and speakers. The matter becomes extremely difficult when tests on combined A. M. and F. M. sets are wanted.

LECTURER: It is agreed that there is difficulty. The B.B.C. is anxious, however, that its programmes should be good enough to be listened to with wide-range loudspeakers.

KNOWLES, H.: Where there are competing broadcasting facilities one must examine the question of whether the station output should favour the overwhelming number of listeners who cannot afford an expensive receiver at least for the programmes of general interest. For this reason in the U.S. some stations use more nearly an average response for monitoring.

LECTURER: The B.B.C. endeavours to give a service which will be satisfactory on the better class of receiver.

SCHIESSER, H. O.: Concerning loudspeaker tests good results were obtained with a method of comparing the sound produced by a natural source of white noise (air pistol under pressure) with the same sound reproduced by the loudspeaker under test. The loudspeaker under test gets its signal from a dead room, where a second equal source of white noise and microphone are located. The first source and loudspeaker are working in the testing room close together. Both sound sources can be compared from time to time.

LECTURER: White noise is frequently used in making tests on loudspeakers, and has proved to be an excellent way of judging colourations subjectively.

APPLICATION OF THE RECIPROCITY THEOREM TO MAGNETIC REPRODUCING HEADS

by W. K. WESTMIJZE

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Summary. It is shown that the output of a magnetic reproducing head can be calculated if the field distribution in front of the head is known for the case when the latter is energized. Application of this method enables us to predict the response curve of a wide-gap head and to explain the existing difference between wide-gap and narrow-gap measurements.

Sommaire. On montre comment on peut calculer le flux magnétique dans la bobine d'une tête reproductrice connaissant la distribution du champ devant la fente de cette tête excitée. Cette méthode permet de déterminer la réponse en fréquence d'une tête à fente large et d'expliquer les différences existant entre les résultats trouvés avec une fente large et une fente étroite.

Zusammenfassung. Es wird gezeigt, wie der magnetische Fluß in der Spule eines Wiedergabekopfes berechnet werden kann, wenn die Feldverteilung vor dem Spalt des erregten Kopfes gegeben ist. Die Anwendung dieser Methode ermöglicht es, die Frequenzkennlinie eines Wiedergabekopfes mit breitem Spalt zu bestimmen und die zwischen Messungen bei breitem und bei schmalem Spalt bestehende Differenz zu deuten.

In magnetic recording there is no way of measuring optically the flux recorded in the tape. There is in fact no method of measuring this quantity with sufficient accuracy other than by using a reproducing head.

If, therefore, there is question of standardizing the recorded flux, as there is at present with regard to the international exchange of radio programmes, it is important to know the physical behaviour of a reproducing head in contact with a magnetized tape. For only by knowing this can the flux in the head become an indication of the flux recorded in the tape. The aim of the present contribution is to illustrate a new method of approach to this problem. This method enables to give an explanation of the observed discrepancy between the so-called "wide-gap" and the "narrow-gap" method of measuring the recorded flux.

We will consider the tape from a macroscopic point of view and assign to each element of the tape a certain magnetization M (A/m). For reasons of simplicity, we will assume that the magnetization is

longitudinal. For a sinusoidally magnetized tape we may accordingly write $M_x = M_0 \cos(2\pi x/\lambda)$, where x is the coordinate along the direction of movement of the tape. Dividing the tape in small slices of length dx, the magnetic moment of each slice is $MA\,dx$, where A is the surface of the cross-section.

In its exterior action, each slice is equivalent with a current around its contour of strength $dI = M dx$. In order to calculate the flux in a reproducing head we may accordingly replace the tape by a series of currents in closed circuits around the tape.

Now we can apply the reciprocity theorem. This may be formulated as follows: if a current I in a coil 1 produces a flux Φ through a coil 2, then an equal current I in coil 2 will produce an equal flux Φ in coil 1. This is equivalent in fact to the statement that the coefficient of mutual induction is the same.

All that we need know, therefore, is the flux through the imaginary coils round the tape caused by a current in the reproducing coil. This flux distribution may be obtained either by measurement or by calculation. The sum of the flux contributions of each of these imaginary coils, energized with a current through the head conformable to the magnetization of the tape, gives the desired flux in the reproducing head. We shall express this in mathematical form.

Let the magnetic potential in front of the head, caused by a current of unit strength through the coil of the reproducing head, be given by $V(x, y)$. The field strength along the x-direction is $H_x = -\partial V/\partial x$, and the flux in a cross-section of the tape of width b and height dy:

$$d\Phi = -\mu_0 \frac{\partial V}{\partial x} b\,dy \quad (\mu_0 = 4\pi \cdot 10^{-7}\ \mathrm{Vs/Am}).$$

A current $M_0 \cos(2\pi x/\lambda)\,dx$ round this cross-section gives accordingly a flux contribution in the reproducing coil equal to

$$d\Phi = -\mu_0 M_0 \cos 2\pi \frac{x}{\lambda} \cdot \frac{\partial V}{\partial x} b\,dx\,dy,$$

and the total flux is obtained by integration over the length and the thickness of the tape. If the thickness d of the tape is so small that $\partial V/\partial x$ is constant over this interval, then the total flux is

$$\Phi = \Phi_0 \int_{-\infty}^{+\infty} \frac{\partial V}{\partial x} \cos 2\pi \frac{x}{\lambda}\,dx,$$

where Φ_0 stands for $\mu_0 M_0 bd$, the amplitude of the recorded flux variation in the tape. The function $\partial V/\partial x$ differs only measurably from zero in the neighbourhood of the gap, say at $x = x_0$. Obviously Φ is a function of x_0 and, $\partial V/\partial x$ being an even function of x for reasons of symmetry, Φ attains its maximum value for gap positions $x_0 = 0, \lambda, 2\lambda, \cdots$. This maximum value is the amplitude of the reproduced flux variation.

For some special potential distributions the calculation of this integral is very simple.

1. Let the tape be in close contact with a reproducing head having an infinitely short gap at $x = x_0$. At the gap the potential along the surface of the head makes a jump 1 (Fig. 1a), and therefore $\partial V/\partial x = \delta(x - x_0)$. The flux in the reproducing coil is $\Phi = \Phi_0 \cos(2\pi x_0/\lambda)$ which represents the well-known fact that, if the gap is short compared with the wavelength, the head reproduces the flux in front of the gap.

2. Another simple case is that in which the potential rises linearly over a length l (Fig. 1b).

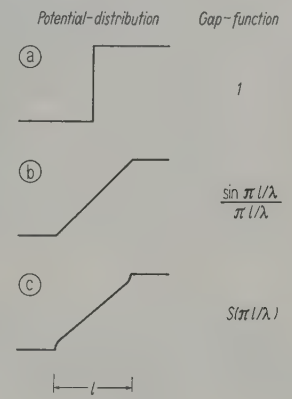


Fig. 1.

This gives $\partial V/\partial x = 1/l$ over the interval $x_0 - l/2 < x < x_0 + l/2$, and $\partial V/\partial x = 0$ outside this interval. The flux is found to be

$$\Phi = \Phi_0 \cos \frac{2\pi x_0}{\lambda} \frac{\sin \pi l/\lambda}{\pi l/\lambda}.$$

This is the well-known gap-loss formula, deduced for the first time by LÜBECK [1], and used for the evaluation of the wide-gap measurements. It will be clear, however, that the potential distribution in front of a gap is not as sketched above and that this formula will only give a rather rough approximation.

The exact distribution of the potential in front of a wide gap (Fig. 1c) can be found with a CHRISTOFFEL-SCHWARZ transformation. This has been worked out in more detail in a recent publication [2], and it leads to a rather complicated gap-loss function, represented by

$$S(\pi l/\lambda) = \frac{1}{\pi} \int_0^{\pi/2} \tan x \sin\left[\frac{2l}{\lambda}\left(\tan x - x + \frac{\pi}{2}\right)\right] dx +$$

$$+ \frac{1}{\pi} \sin \frac{\pi l}{\lambda} \int_0^1 \frac{x}{1-x^2} \exp\left[\frac{2l}{\lambda}\left(x + \frac{1}{2} \ln \frac{1-x}{1+x}\right)\right] dx.$$

This formula holds for a tape running in close contact with the reproducing head. The mathe-

matical analysis shows that, for a (thin) tape running with a spacing a between tape and head, this formula has to be multiplied by $\exp(-2\pi a/\lambda)$.

The open voltage over the coil of a reproducing head in contact with a thin tape is given by

$$E = 2\pi \frac{v}{\lambda} S(\pi l/\lambda) \Phi_0$$

where v is the tape velocity. The function $(\pi l/\lambda) \cdot S(\pi l/\lambda)$ is given in Fig. 2. This figure shows that the maxima of this function rise with respect to the

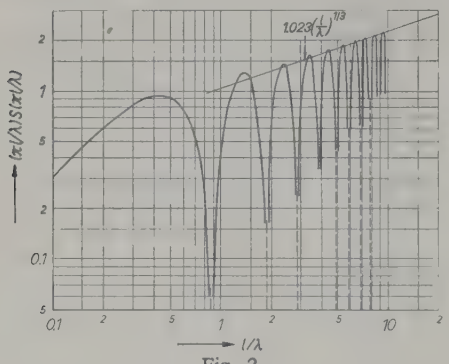


Fig. 2.

function $\sin(\pi l/\lambda)$, with a factor $(l/\lambda)^{1/3}$, or 2 dB per octave, while the zeros occur at somewhat different values. The first zero occurs at $l/\lambda = 0.88$. This is in excellent agreement with experiments, especially recent measurements of DANIEL and AXON [3] who found experimentally $l/\lambda = 0.87$.

Although the reciprocity theorem gives a simple method of evaluating the flux in the reproducing head, the flux can of course also be calculated by other methods. A recent publication of SCHMIDBAUER [4] gives a solution where, in an approximation for small wavelengths the 2 dB/oct. is also obtained.

A few words should be said here about the validity of the gap-function deduced above. In the derivation it was tacitly assumed that the permeability was unity. If this is not so the distribution of the magnetization is no longer sinusoidal if the tape is in contact with the head, the demagnetization being stronger in front of the gap than in front of the iron of the head. Measurements with different tapes show, however, that the influence of the permeability is less than 0.5 dB.

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SECTION II

ELEKTROAKUSTISCHE ÜBERTRAGUNGSANLAGEN

(PUBLIC ADDRESS SYSTEMS)

HAUPTBERICHT

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1. Einleitung

Die elektroakustischen Übertragungsanlagen haben in vieler Hinsicht das Leben unserer Zeit beeinflußt. Vorträge in großen Räumen, Massenversammlungen in Hallen und unter freiem Himmel, Sportveranstaltungen und dgl. sind erst durch den Einsatz von elektroakustischen Anlagen möglich geworden. Die eben aufgezählten Beispiele zeigen, daß es im wesentlichen die menschliche Sprache ist, die der Verstärkung bedarf, um auf größere Entfernungen klar und mühelos verstanden zu werden. Demgegenüber tritt die Anwendung elektroakustischer Anlagen auf musikalischem Gebiet (vom Rundfunk und Tonfilm darf im vorliegenden Zusammenhang abgesehen werden) erheblich zurück. Aber es ist gerade ein gewisses Kennzeichen der Entwicklung der letzten Zeit, daß auch hier die Elektroakustik durch die laufende Verbesserung der Qualität ihrer Anlagen und durch die Ausnutzung psychoakustischer Effekte im Begriff ist, einen Vorstoß in das bisher ihr aus künstlerischen Gründen verschlossene Gebiet der unmittelbaren Verstärkung von musikalischen Aufführungen zu machen.

Das Gebiet der elektroakustischen Übertragungsanlagen ist außerordentlich umfangreich. Wir werden zunächst allgemeine Gesichtspunkte wie Verzerrungen und Richtwirkung besprechen und dann auf die drei wesentlichen Organe: Mikrophon, Verstärker und Lautsprecher eingehen. In einem abschließenden Teil werden wieder allgemeine Fragen, besonders die Aufstellung der Lautsprecher und der Einfluß der Raumakustik, herausgestellt.

2. Verzerrungen

Die akustischen Qualitätsansprüche an Mikrophone, Verstärker und Lautsprecher sind die gleichen wie auf den anderen Gebieten der Elektroakustik und hinreichend bekannt, so daß eine Aufzählung genügt. An erster Stelle steht die Forderung nach einer genügenden Frequenzbandbreite. Gewünscht wird immer wieder der gesamte Frequenzumfang des normalen Gehörs. Praktisch wird der Frequenzbereich jedoch durch den technischen Aufwand bestimmt, obwohl kein Zweifel daran besteht, daß ein großer Frequenzbereich mit Rücksicht auf die Musik unter sonst guten Übertragungsbedingungen die besten Ergebnisse liefert. Kommt es nur auf die Verständlichkeit von Sprache an, so kann man natürlich davon abgehen, und es ist mehr eine Frage des Geschmacks, wie weit man den Frequenzbereich einengen will. In halligen Räumen, beispielsweise in Kirchen, macht man sehr oft davon Gebrauch, die tiefen Frequenzen mehr oder weniger abzuschwächen, da sie für die Verständlichkeit unerheblich sind, dagegen durch ihre meist große Nachhallzeit den allgemeinen Geräuschpegel erhöhen. Mit der Beschränkung der hohen Frequenzen muß man dagegen vorsichtiger sein. Man kann sie zwar, wie die Telephonie beweist, entbehren, aber ein leichtes und müheloses Verstehen komplizierten Textes erfordert die Übertragung bis etwa 8000 Hz. Erwähnt sei auch, daß die Frage, welche obere Grenzfrequenz bei Lautsprecheranlagen mit Laufzeitverzögerung günstig ist, noch nicht genügend geklärt wurde; es findet sich der Hin-

weis, daß der Frequenzbereich bis 4000 Hz genügt [1].

Hinsichtlich eines großen Frequenzumfanges macht das Mikrophon gar keine Schwierigkeiten. Mikrophone verschiedener Typen stehen zur Verfügung, deren Frequenzkurven bei genügender Empfindlichkeit allen Ansprüchen entsprechen. Das gleiche gilt für den Bauteil Verstärker, dessen Konstruktion wegen des Fehlens elektromechanischer Glieder noch einfacher zu beherrschen ist. Das schwächste Glied in der Übertragungskette ist der Lautsprecher. Überblickt man die zahlreichen Entwicklungen der beiden letzten Jahrzehnte auf diesem Gebiet, so folgt daraus, daß es nach dem derzeitigen Stand der Kenntnisse unmöglich ist, den gesamten Frequenzbereich durch ein einziges Lautsprecher-System wiederzugeben. Eine Kombination von Systemen, auch wenn sie rein äußerlich zu einer Einheit verschmolzen sind, ist immer erforderlich.

Mit dem Problem der Frequenzkurve ist aufs engste die Frage der Phasenkurve oder auch anders ausgedrückt, die Frage der Einschwingzeiten gekoppelt. Man weiß, daß in der Sprache, in den Konsonanten oder Übergangslauten von Konsonanten zu Vokalen, ganz kurze Wellengruppen in der Größenordnung von 1 ms Dauer vorkommen. Dem dürfen die elektrischen oder elektromechanischen Teile einer Übertragungsanlage nichts hinzufügen. Einfach liegen die Dinge wieder beim Verstärker und beim Mikrophon, schwierig dagegen beim Lautsprecher. Die häufig zu beobachtende „topfige" oder sonst irgendwie unnatürlich klingende Sprachwiedergabe geht zum überwiegenden Teil auf die Einschwingvorgänge des Lautsprechers zurück. Die Forderung nach Freiheit von Einschwingvorgängen ist aber im Gegensatz zu früher vielfach geäußerten Ansichten identisch mit einer geraden Frequenzkurve in dem betreffenden Frequenzbereich.

K. W. WAGNER und auch J. PETERS [2] haben gezeigt, daß Phasenlaufzeit und Frequenzkurve des Betrages unmittelbar miteinander verknüpft sind, wobei im wesentlichen die Phasenlaufzeit durch die Steigung der Frequenzkurve (aufgetragen über dem Logarithmus der Frequenz) gegeben ist.

In diesem Zusammenhang darf an ältere Versuche von W. BÜRK, P. KOTOWSKI und H. LICHTE [3] erinnert werden, die für zwei rechteckig geschaltete Sinusschwingungen die eben wahrnehmbare Laufzeitdifferenz festgestellt haben; die Differenz ist im Frequenzbereich 200···4000 Hz besonders klein (kleiner als 10 ms).

KÜPFMÜLLER [4] hat gezeigt, daß des Einschwingvorganges wegen eine Frequenzkurve an den Eckfrequenzen nicht steil abfallen darf, sondern einen cos-artigen Verlauf haben muß. Der günstigste Frequenzverlauf in der Nähe der oberen (f_2) und der unteren Grenzfrequenz (f_1) ist durch die Funktion

$$20 \log \frac{2}{1 + \cos(\pi f/f_2)} \text{ dB}$$

bzw.

$$20 \log \frac{2}{1 - \cos(\pi f/f_1)} \text{ dB}$$

gegeben.

Der zweite Punkt im Abschnitt „Verzerrungen" ist die nichtlineare Verzerrung. Methoden, sie zu messen, sind in großer Zahl bekannt. In dem letzten Jahrzehnt hat sich dabei das schon sehr früh, meines Wissens zum ersten Mal von E. WAETZMANN [5] beim Kohle-Mikrophon angewendete Zwei-Ton-Verfahren ein umfangreiches Anwendungsgebiet verschafft. Gemessen werden dabei in der Regel die durch die nichtlineare Verzerrung entstehenden Modulationsprodukte zweier sinusförmiger Primärfrequenzen, wobei man zwischen quadratischer und kubischer Verzerrung zu unterscheiden hat. Die wichtigste Frage ist, welche Verzerrungen gerade noch zugelassen werden können; sie ist analog der Frage nach dem maximalen Frequenzumfang schwer zu beantworten. Es besteht kein Zweifel, daß in dieser Hinsicht eine gute Wiedergabe von Sprache geringere Ansprüche als eine naturgetreue Wiedergabe von Musik stellt. Zur Frage der nichtlinearen Verzerrungen sind in den letzten Jahren z.B. Untersuchungen von FELDTKELLER und seinen Schülern [6] angestellt worden. Er fand an Musik und hier speziell an Quintenintervallen, daß Klirrfaktoren von weniger als 1% bis 0,5% noch bemerkbar sind. Derartig geringe Werte für das Mikrophon zu erreichen, bereitet keine Schwierigkeiten; größere Bemühungen verlangen dagegen Endverstärker und Lautsprecher.

Der Vollständigkeit wegen sei noch auf den dritten Punkt hinsichtlich der „Verzerrungen" hingewiesen, der zu Beschwerden Anlaß gibt, nämlich den aus einer Übertragungsanlage kommende Geräuschpegel. Im wesentlichen ist dies allerdings eine Frage der rein elektrischen Geräte; sie betrifft die Abschirmung gegen äußere elektromagnetische Störfelder, die Beseitigung von Netzstörungen und das Kleinhalten des Widerstands- oder Röhrenrauschens am elektrischen Eingang der Übertragungskette.

3. Richtwirkung

Eine große Bedeutung hat, besonders mit Rücksicht auf die raumakustischen Gegebenheiten, die Richtwirkung der elektroakustischen Anlage, sowohl was den Schallempfang als auch die Schallabstrahlung betrifft. Aber auch für größere Wiedergabeanlagen im Freien bedeutet eine Richtwirkung einen Gewinn an Leistung. Mit dem gerichteten Schallempfang und der gerichteten Schallwiedergabe hat man sich schon sehr früh in der Elektroakustik befaßt. Leider ist es grundsätzlich so, daß man eine hohe Richt-

wirkung ohne Einbuße an Empfindlichkeit nur durch Anlagen mit großer räumlicher Ausdehnung erreicht. Für Mikrophone kann man zwar durch wiederholte Anwendung des Gradient-Prinzips bei kleiner räumlicher Ausdehnung beliebig scharfe Richtcharakteristiken erzielen; leider wird aber auch die Empfindlichkeit dabei beliebig klein. Mikrophone in Parabolspiegeln oder das Linien-Mikrophon sind ältere Beispiele gerichteter Empfänger. Trichter-Lautsprecher mit großen Öffnungsflächen sind oft als gerichtete Strahler verwendet worden.

Zur Zeit arbeitet man fast ausschließlich nach dem Prinzip der linearen Gruppe, wie es aus der Wasserschalltechnik schon seit Jahrzehnten bekannt ist [7]. Eine lineare Gruppe von Einzelstrahlern oder Einzelempfängern hat die Eigenschaft, daß sie senkrecht zur Gruppe stark gerichtet abstrahlt oder empfängt. Die Richtwirkung hängt von ihrer Ausdehnung sowie dem Abstand der Einzelstrahler oder Einzelmikrophone voneinander im Verhältnis zur Wellenlänge ab. Abb. 1 zeigt ein Beispiel für eine

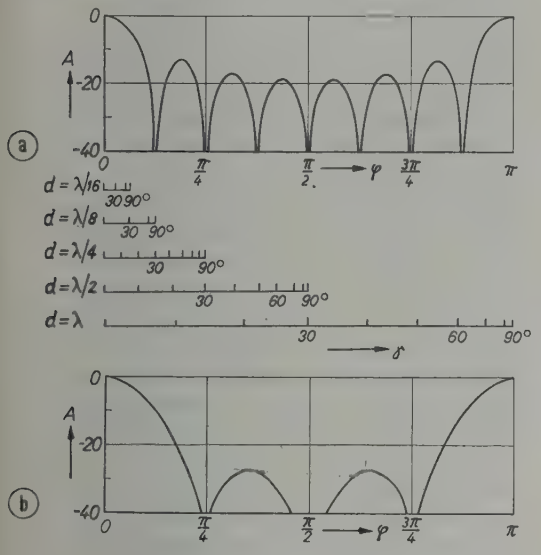


Abb. 1. Richtcharakteristiken von Zeilen mit acht Systemen (Länge d; $\varphi = \frac{\pi d}{\lambda} \sin \gamma$).

a) alle Systeme mit gleicher Empfindlichkeit,
b) Empfindlichkeit, die nach beiden Seiten der Zeile linear bis auf Null abnimmt.

Gruppe von acht Systemen. Als Ordinate ist in logarithmischem Maß die Amplitude, bezogen auf das Hauptmaximum, als Abszisse der Ausdruck $\varphi = (\pi d/\lambda) \sin \gamma$ bzw. für verschiedene Verhältnisse d/λ der Winkel γ aufgetragen, unter dem der Schall aufgenommen oder abgestrahlt wird.

Wie bei jeder Richtanlage wird der Gewinn mit wachsender Frequenz immer größer, d.h. der Richteffekt immer schärfer, und bei diskontinuierlich verteilten Einzelsystemen treten außer dem gewünschten Hauptmaximum senkrecht zur Gruppe noch weitere Hauptmaxima mehr oder weniger schräg zur Gruppe auf. Beide Effekte führen dazu, daß man die Richtwirkung für höhere Frequenzen abschwächen muß. Bei Lautsprecherzeilen tut man dies z.B. dadurch, daß man für Frequenzen unterhalb von 1000 Hz eine größere Gruppe, für diejenigen oberhalb 1000 Hz eine kleinere Gruppe benützt. Die größte in der Literatur bisher bekannt gewordene Lautsprecherzeile ist etwa 7 m lang.

Bei Mikrophonzeilen, die z.B. aus Kondensator-Mikrophonen bestehen, setzt man die mittleren Mikrophone dichter und die äußeren Mikrophone weiter auseinander und sorgt außerdem durch die elektrische Schaltung dafür, daß mit wachsender Tonhöhe die äußeren Mikrophone unwirksam werden. Auf diese Weise erhält man eine einigermaßen frequenzunabhängige Richtwirkung.

Bekannt ist auch der Kunstgriff zur Vermeidung der kleineren Nebenmaxima, die Empfindlichkeit der äußeren Systeme einer Gruppe kleiner zu machen als die der mittleren Systeme (Abb. 1b) [8].

Um Lautsprecher nur in einen Halbraum abstrahlen zu lassen, bringt man nach dem Vorschlag von SPANDÖCK [9] auf der Rückseite ein Laufzeitglied (bestehend aus einem gedämpften Resonator) an, und erhält damit eine Art Kardioid-Charakteristik.

In ähnlicher Weise kann man bei den Mikrophonzeilen vorgehen und hierfür allseitig empfindliche Mikrophone, aber auch Gradient- oder Kardioid-Mikrophone benutzen.

Mikrophonzeilen braucht man, wenn es sich um die Aufnahme von sich bewegenden Schallquellen, von Vortragenden in größeren Hörsälen oder von Schauspielern auf Bühnen handelt. Im allgemeinen steht das Mikrophon bei sich nicht bewegenden Personen nahe am Sprecher, schon aus dem Grunde, um fremde Schallvorgänge, Störgeräusche oder dgl. auszuschalten. An Stelle von allseitigen Mikrophonen benutzt man vielfach Gradient- oder Kardioid-Mikrophone.

Ist der Geräuschpegel sehr hoch, was im allgemeinen seltener vorkommt, so wird das Mikrophon direkt an den Mund des Sprechers gebracht; verwendet man dabei ein Gradient-Mikrophon, so ist zu beachten, daß der Druckgradient bei Annäherung an eine Kugelschallquelle viel stärker zunimmt als der Schalldruck selbst. Dies macht sich bei tiefen Frequenzen besonders bemerkbar und muß elektrisch entsprechend kompensiert werden.

Der Vollständigkeit wegen sei in diesem Zusammenhang noch das sogenannte Knopfloch-Mikrophon erwähnt, das Personen, die beim Vortrag hin- und hergehen müssen, z.B. auf dem Rockaufschlag

tragen. Da bei dieser Mikrophonlage gegenüber dem Mund des Sprechers die hohen Frequenzen benachteiligt werden, ist auch hier eine entsprechende elektrische Kompensation nötig.

4. Aufnahme-Mikrophon

Im Laufe der Zeit sind sämtliche aus der Physik her bekannten Möglichkeiten zur Umwandlung mechanischer Schwingungen in Wechselspannungen verwendet worden. Erwähnenswert sind hier Versuche von H. HECHT und F. A. FISCHER, die bisher bekannten Wandlerarten systematisch zu ordnen [10]. Vorzugsweise werden jedoch heutzutage das elektrostatische, das elektrodynamische und das piezo-elektrische Mikrophon benutzt. Dabei hat das elektrodynamische Mikrophon mannigfache Vorteile gegenüber allen anderen Mikrophontypen, (z.B. kein Vorverstärker, geringer innerer Widerstand, robuste Konstruktion, weitgehende Unempfindlichkeit gegen Wärme und Feuchtigkeit). Es erübrigt sich, hier auf einzelne Konstruktionen einzugehen. Membran-Mikrophone, z.B. in der von WENTE-THURAS angegebenen Bauweise [11], und Band-Mikrophone, erstmalig wohl von GERLACH [12] eingeführt, sind im Gebrauch.

Mikrophone sind in der Regel allseitig empfindlich, bzw. besitzen entsprechend ihrer Größe nur für die hohen Frequenzen einen gewissen Richteffekt. Interessant ist eine in Deutschland viel benutzte und fast ungerichtet wirkende Konstruktion „Wennebostel", bei der das Mikrophon über eine enge Rohrleitung besprochen wird, deren verstellbare Eingangsöffnung sich in Mundhöhe befindet und deren anderes Ende in einem Stativfuß das eigentliche Membran-Mikrophon, akustisch möglichst gut angepaßt, trägt (Abb. 2) [13].

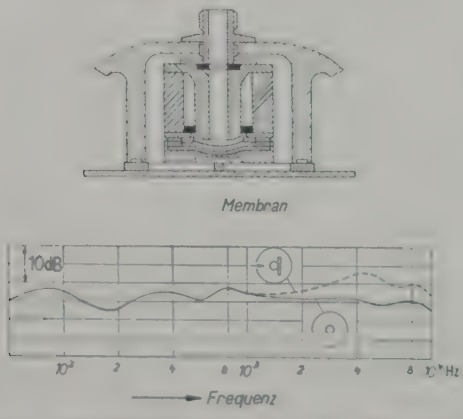


Abb. 2. Rohrmikrophon vom Laboratorium Wennebostel.

Das Band-Mikrophon ist in dem letzten Jahrzehnt von H. F. OLSON [14] zu großer Vollkommenheit entwickelt worden. Es besitzt zum Teil sogar eine veränderbare Richtcharakteristik (Kugel-, Achter- oder Nieren-Charakteristik). Auch beim Band-Mikrophon sind Konstruktionen bekannt geworden, die optisch unauffällig sind [15] (Abb. 3).

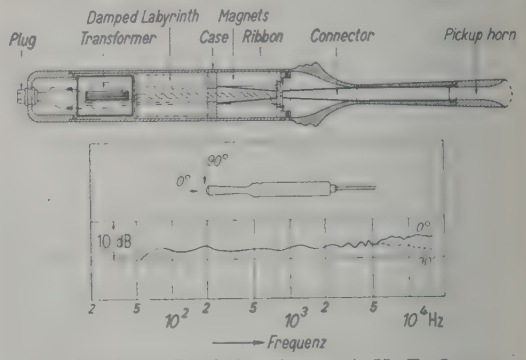


Abb. 3. Kleines Bandmikrophon nach H. F. OLSON.

Elektrostatische Mikrophone werden heute vielfach mit einer dünnen, metallbelegten Kunststoffmembran hergestellt. Besonderer Wert wird dabei auf einen kleinen Vorverstärker gelegt, der sich unmittelbar an das Mikrophon anschließt. Das elektrostatische Mikrophon besitzt gegenüber anderen Typen den Vorteil, daß es ohne Schwierigkeiten in seinem Frequenzbereich bis weit über die menschliche Hörgrenze ausgedehnt werden kann. Eine besondere Stellung nehmen in dieser Beziehung die Mikrophone mit festem Dielektrikum nach SELL [16] ein, d.h. Kondensator-Mikrophone, bei denen eine metallbedampfte, sehr dünne Kunststoff-Folie unmittelbar auf der rückwärtigen Gegenelektrode aufliegt. Man erreicht je nach deren Ausbildung (glatte Oberfläche, sandgestrahlte Oberfläche oder mit schmalen Rillen geringer Tiefe versehenen Oberfläche), was allerdings im vorstehenden Zusammenhang ohne Bedeutung ist, eine große Empfindlichkeit gekoppelt mit einer Frequenzunabhängigkeit bis zu 100 kHz.

Die nachfolgende Tafel stellt die Durchschnittswerte der Empfindlichkeit (dB über 1 V/μbar *) und

Mikrophon	Empfindlichkeit (dB über 1 V/μbar)	Anpassung an Widerstand von	Grenzschalldruck (über $2 \cdot 10^{-4}$ μbar)	Grenzlautstärke
Tauchspulen-Mikrophone	−80 dB	200 Ω	19 dB	
Rohr-Mikrophone	−86 dB	200 Ω		
Bändchen-Mikrophone	−80 dB	200 Ω		
Kondensator-Mikrophone im Hörbereich	−54 dB	Leerlauf	24 dB	20 phon

*) 10 μbar = 1 N m².

der inneren Widerstände von einigen Mikrophontypen zusammen.

Nach kleinen Schalldrucken zu ist die Schallaufnahme jedes Mikrophons durch die eigenen Störgeräusche begrenzt, die z.B. durch das thermische Rauschen der Wirkkomponente des Eingangswiderstandes gegeben sind. Eine wichtige Größe zur Charakterisierung der Mikrophone ist daher ihr sogenannter Grenzschalldruck oder ihre Grenzlautstärke. Das ist die Störspannung oder lautstärkemäßig bewertet die Geräuschspannung im Eingangskreis dividiert durch das Übertragungsmaß des Mikrophons. Es ist die für ein Mikrophon im eigentlichen Sinne wesentliche Größe. Man kennt Kondensator-Mikrophone mit einer Ersatzlautstärke bis zu 20 phon, wenn sie in einer Niederfrequenzschaltung benutzt werden. Kondensator-Mikrophonen in Hochfrequenzschaltungen wird eine sehr kleine Ersatzlautstärke nachgerühmt.

5. Lautsprecher

Wenn beim Mikrophon verschiedenartige Systeme verwendet werden, so ist beim Lautsprecher festzustellen, daß hier uneingeschränkt das elektrodynamische System herrscht. Es kommt zwar vor, daß man Zusatzlautsprecher in Rundfunkgeräten und dgl. in Form von elektrostatischen Systemen, z.B. nach dem schon oben erwähnten Sellschen Prinzip verwendet. Aber in die hier zur Diskussion stehenden Übertragungsanlagen wurden sie bisher nicht eingebaut. Auch die Schallabstrahlung durch „ionisierte" Gasstrecken ist des öfteren versucht worden.

Es ist das Rice-Kelloggsche System der tief abgestimmten, mehr oder weniger konusförmigen Pappenguß-Membran, das ausschließlich im Bereich der tiefen und mittleren Frequenzen benutzt wird. Hierbei sind in den beiden letzten Jahrzehnten viele wesentliche Verbesserungen angebracht worden. Bestimmte flache Konusformen erwiesen sich als günstig, da sie bei starker Erregung keine Untertöne erzeugen (Nawi-Membran) (Buchmann [4]). Die störenden Resonanzen der Sicken, d.h. der nachgiebigen Randteile der Membran, werden durch Überziehen mit einem zäh bleibenden Kunststoff gedämpft [4]. Von japanischer Seite (T. Nimura, E. Matsui, K. Shibayama und K. Kido [17]) sind kürzlich eingehende theoretische und experimentelle Studien über die Schwingungsmöglichkeiten des Konus gemacht worden, wobei sowohl Dehn- wie Biegeschwingungen berücksichtigt sind; die mechanische Nachgiebigkeit am inneren Konusrand spielt die Hauptrolle.

Auch Vorschläge für zwei ineinander geschachtelte Tiefton- und Hochton-Konusse oder Vorschläge für einen einzigen großen, aber durch biegeweiche Zwischenzonen in verschiedene Frequenzbereiche aufgeteilten Konus wurden gemacht [18].

Interessant sind kürzliche Veröffentlichungen von Corrington und Kidd [19] über die Schwingungsformen eines Lautsprecher-Konus, die mit elektrostatischer Abtastung bestimmt wurden. Auch hier zeigt sich, daß eine starke Resonanz der Randpartien auftritt; für höhere Frequenzen sind erhebliche Schwingungsunterteilungen des Konus vorhanden.

Im Gegensatz zu den Trichter-Lautsprechern bedürfen, wie bekannt, die Konus-Lautsprecher eines besonderen Einbaues, um den Strahlungswiderstand bei tiefen Frequenzen anzuheben. Auch ein ebener Schallschirm verändert durch die Schallbeugung an seinem Rande die Frequenzkurve, wie schon G. Buchmann 1936 [20] gezeigt hat. Ähnliche Versuche mit sehr verschiedenartigen Umbauten des Konussystems wurden kürzlich von H. F. Olson [21] gemacht, der gleichfalls den erheblichen Einfluß der Umrandung nachwies. Das Verfahren, durch Anbau eines gedämpften Resonators als Laufzeitglied auf der Rückseite des Konus die abgestrahlte Leistung nach vorn (Kardioid-Charakteristik) zu erhöhen, wurde bereits im Abschnitt „Richtwirkung" erwähnt. Über die Größe des Schallschirmes geben Nimura und Watanaba [22] an, daß der Durchmesser eines kreisförmigen Schirmes etwa gleich der halben Wellenlänge der noch gewünschten tiefen Frequenz sein soll.

Trichter-Lautsprecher sind zwar theoretisch ideal, weil eine besonders kleine und damit leichte Membran verwendet werden kann, die unter Transformation des akustischen Strahlungswiderstandes durch eine Druckkammer akustisch stark belastet und gedämpft wird. Insbesondere ist damit, wie bekannt, eine vorzügliche Abstrahlung der tiefen Töne erzielbar. Zwei Gründe verhindern jedoch die praktische Anwendung des Trichter-Lautsprechers in größerem Umfange; einmal die unhandliche Größe des Trichters, der mit abnehmender Grenzfrequenz immer größer werden muß, und zudem macht für die hohen Frequenzen die Führung der Schallwellen im Trichter Schwierigkeiten, d.h., es bilden sich in ihm stehende Querwellen aus. Man verhindert deren Auftreten weitgehend durch eine starke Unterteilung des Trichters in viele kleine Einzeltrichter. Begnügt man sich, wie zur Wiedergabe von Sprache, mit einer höheren Grenzfrequenz, so ist der Trichter-Lautsprecher sehr brauchbar und wird mit einem Falttrichter viel angewendet; insbesondere fällt an ihm die klare Sprachwiedergabe auf.

Sehr kleine Trichter-Lautsprecher werden häufig als Zusatzlautsprecher für die hohen Frequenzen benutzt.

Zahlreiche Diskussionen sind über die Messung der Frequenzkurve von Lautsprechern entstanden. Man kann in einem reflexionsfreien Raum den Schalldruck eines Lautsprechers in verschiedenen Richtungen aufnehmen und erhält damit zu den Frequenzkurven die Richtdiagramme. Man kann auch bei der Aufnahme das Mikrophon, wie kürzlich vorgeschlagen wurde, in einem gewissen Bereich, z.B. längs eines Kreises, hin- und herbewegen und erhält dabei einen Mittelwert (J. C. HENTSCH [23]). Ein anderes schon sehr früh angegebenes Verfahren besteht darin, die gesamte abgestrahlte Intensität des Lautsprechers in einem Hallraum zu messen (E. MEYER [24]); eine solche Messung zeigt übrigens, daß bei allen Lautsprechern die gesamte abgestrahlte Energie nach den hohen Frequenzen zu erheblich abnimmt. Diese Erkenntnis hat in jüngster Zeit dazu geführt, daß man für die Wiedergabe der Frequenzen über 400 Hz einen aus vielen Einzelstrahlern zusammengesetzten Kugelstrahler entwickelt hat (KÖSTERS [25]). Nach allen Seiten abstrahlend ist er elektrisch so entzerrt, daß er bei gleicher Erregung für alle Frequenzen die gleiche Strahlungsleistung abgibt. Die Folge dieser Maßnahmen ist übrigens ein unerhört räumlicher Eindruck. Es ist unmöglich, auch bei ganz kurzer Entfernung, den Lautsprecher präzise zu lokalisieren.

Was die Frequenzkurve von Lautsprechern betrifft, so kann man natürlich durch eine Kombination mehrerer Lautsprechersysteme und durch entsprechende elektrische Entzerrungen jede gewünschte Kurve herstellen.

Die nichtlineare Verzerrung ist durch Verwendung eines elektrodynamischen Antriebssystems geeigneter Konstruktion relativ gering. Man findet in der Literatur z.B. Angaben, daß die zweite Harmonische bei 10 Watt Leistung etwa 2% und die dritte Harmonische 1% beträgt.

Erwähnenswert ist noch die Verzerrung durch den Dopplereffekt. Führt ein Lautsprecher bei tiefen Frequenzen sehr große Amplituden aus und strahlt dabei gleichzeitig hohe Frequenzen ab, so zeigen diese wegen der Bewegung der Schallquelle eine Phasenmodulation, die zu Verzerrungen Anlaß gibt. Man vermindert diesen Effekt dadurch, daß man die Wiedergabe der hohen und der tiefen Frequenzen verschiedenen Systemen zuweist, was man auch aus anderen Gründen, wie oben erwähnt, anstrebt.

Ein besonders einfaches Verfahren, Einschwingvorgänge von Lautsprechern zu messen, besteht darin, sie mit einer rechteckförmigen Stromkurve zu erregen.

Eine genauere, schon vor langer Zeit angewandte Methode (HELMBOLD [26]) Einschwingvorgänge in Abhängigkeit von der Frequenz zu messen, benutzt die Ein- und Ausschaltung von Sinusspannungen. Je nachdem, ob die Schaltfrequenz mit einem Maximum oder einem Minimum oder einer zwischenliegenden Stelle in der Frequenzkurve zusammenfällt, steigt der gemessene Schalldruck langsam an, oder er steigt sehr schnell an und überschreitet den stationären Wert, um dann auf diesen abzufallen, oder er behält annähernd die Form der Einschaltkurve. Bezeichnet man in den drei genannten Fällen die Einschwingzeit als positiv, negativ oder Null, so findet man Kurven wie in Abb. 4, die auch die Amplitudenkurven wiedergibt.

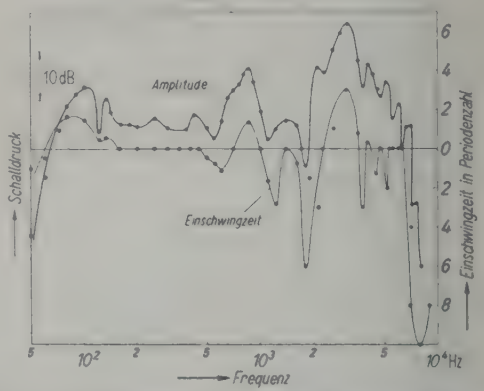


Abb. 4. Amplitudenkurve und Einschwingzeitkurve eines elektrodynamischen Lautsprechers nach HELMBOLD.

Eine neuere Methode ist von EWASKIO und MAWARDI [27] angegeben worden. Die zeitliche Verzögerung der Hüllkurve oder Phasenverschiebung einer amplitudenmodulierten Sinusschwingung (Modulationsfrequenz sehr tief) wird in Abhängigkeit von der Trägerfrequenz gemessen. Beispiele hierfür zusammen mit den Amplitudenkurven geben Abb. 5a und 5b. Auch diese Abbildungen betreffen, wie

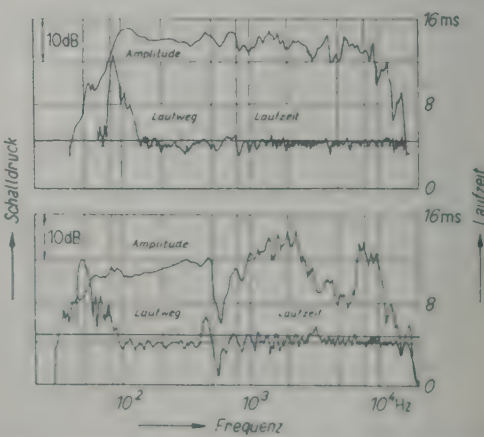


Abb. 5. Amplituden- und Laufzeitkurven von zwei elektrodynamischen Lautsprechern nach EWASKIO und MAWARDI.

Abb. 4, elektrodynamische Konus-Lautsprecher; die konstanten Werte der Laufzeit in Abb. 5 sind durch den Schallweg Lautsprecher—Meßmikrophon gegeben. Bei jeder Änderung der Amplitudenkurve erfolgt die entsprechende Schwankung der Phasenkurve.

Erwähnt sei noch das Iterationsverfahren, das kürzlich MEYER-EPPLER [28] angegeben hat. Es „potenziert" die Verzerrungen einer Übertragungsanlage durch wiederholte Aufnahme und wiederholte Wiedergabe; allerdings ist bei diesem Verfahren erforderlich, daß das Aufzeichnungsverfahren selbst größenordnungsmäßig besser als das zu testende System ist.

6. Verstärker

Wenn man die augenblickliche Situation auf dem Gebiet der Verstärker zusammenfassen will, so kann man es nicht besser tun, als mit der Feststellung, daß die Verstärker für die elektroakustischen Übertragungsanlagen laufend besser in der Qualität, billiger im Preis und kleiner im Aufbau werden. Frequenzkurven mit Schwankungen von weniger als 1 dB im gesamten Hörbereich sind selbstverständlich. Klirrfaktoren in der Größenordnung von $1^0/_{00}$ sind gleichfalls durch Verwendung von Gegentaktschaltungen und geeignete Transformatoren sowie Benutzung der Gegenkopplung erreichbar. Bei dem Stichwort Gegenkopplung sei erwähnt, daß man versucht hat, in den Gegenkopplungskreis auch die mechanische Bewegung der Lautsprechermembran mit Hilfe einer zweiten Wicklung für die Antriebsspule einzubeziehen. Weiterhin ist zu sagen, daß die Störgeräusche, die durch den Netzbetrieb kommen, durch entsprechende Siebmittel beliebig stark unterdrückt werden können. Der Verstärker ist das Glied in der Kette der Übertragungsmittel einer elektroakustischen Anlage, das sich am einfachsten beherrschen läßt.

7. Elektroakustische Anlagen und der Einfluß der Raumakustik [1])

Auf diesem Gebiet sind in den letzten Jahren die größten Fortschritte erzielt worden. Früher hat die Gefahr einer akustischen Rückkopplung zwischen Mikrophon und Lautsprecher in erster Linie die Aufstellung der Lautsprecher bestimmt. Entschied man sich entsprechend den örtlichen Gegebenheiten für eine zentrale Anlage, d.h. für große Lautsprecher in der Nähe der Schallquelle, so wurden sie in der Regel so aufgestellt, daß sie sich zwischen Schallquelle und Zuhörer befanden, d.h. in jedem Fall kam beim Zuhörer der Lautsprecherschall früher als der direkte Schall der Schallquelle (z.B. Sprecher) an. Hat ein Raum eine große Nachhallzeit, oder sind im Freien große Flächen mit Schall zu versorgen, so verwendete man Anlagen mit verteilten Lautsprechern, d.h. viele kleine Lautsprecher geringer Leistung inmitten des Publikums. Ein gutes Beispiel hierfür bildet die elektroakustische Versorgung von Kirchen, wobei ein Extremum St. Paul's Cathedral darstellte, in der noch vor wenigen Jahren etwa 600 Lautsprecher inmitten der Sitzreihen angebracht waren. Bei den verteilten Anlagen ist es selbstverständlich, daß der Schall aus dem Lautsprecher zuerst den Zuhörer erreicht und damit in der Regel für das Gehör eine ganz andere Richtung festlegt, als sie dem optischen Eindruck entspricht.

Von diesem System beginnt man in jüngster Zeit ganz abzugehen. Hierfür sind zwei Gründe verantwortlich. Es ist dies einmal die Erkenntnis, daß zur Übereinstimmung zwischen akustischem und optischem Richtungseindruck der direkte Schallstrahl Schallquelle-Zuhörer zuerst ankommen muß, d.h. der Lautsprecherschall darf frühestens gleichzeitig, soll jedoch besser etwas später beim Beobachter eintreffen. Diese Forderung ist in mittelgroßen Räumen, in denen der direkte Schall auch beim entferntesten Zuhörer noch gut wahrnehmbar ist, leicht zu erfüllen: Die Lautsprecher müssen einen größeren Abstand vom Publikum haben als der Redner oder die sonstige Schallquelle; dabei spielt die Richtung, in der die Lautsprecher vom Zuhörer aus gesehen stehen, eine untergeordnete Rolle. In sehr großen Räumen, in denen viele Lautsprecher aufgestellt werden, ist eine künstliche Laufzeitverzögerung erforderlich, etwa durch Aufnahme und Wiedergabe des Schalles auf einem endlosen Magnetophonband. Versuche dieser Art sind schon alt. FLANZE und GLADENBECK [29] haben erstmalig 1936 unter Verwendung von akustischen Rohrleitungen (Rohrpost-Leitungen) im Post-Stadion Berlin damit gearbeitet. L. CREMER machte Versuche in der Frauenkirche in München. Am eingehendsten wurde der Effekt von HAAS [30] in seiner Göttinger Doktorarbeit studiert, wobei er die Bedingungen, unter denen die Beibehaltung des Richtungseindrucks möglich ist, quantitativ klärte. Aus diesem Grunde wird dieser Effekt in der englischen Literatur häufig als HAAS-Effekt bezeichnet. Abb. 6 zeigt aus einer späteren Arbeit (MEYER-SCHODDER [31]), in der die Messungen unter etwas variierten Bedingungen durchgeführt sind, um wieviel dB bei den verschiedenen Laufzeitdifferenzen der durch den Lautsprecher hervorgerufene Schallpegel am Ort des Zuhörers größer sein darf als der Pegel des zuerst ankommenden Schalles, im allgemeinen also des direkten Schalles, ohne daß man den ursprüng-

[1]) Schallübertragungen auf stereophoner Basis (raumbezügliche Stereophonie) sollen hier nur erwähnt werden, da sie praktisch in den Public Address Systems keine Rolle spielen.

lichen Richtungseindruck verliert. Dabei ist es sogar möglich, daß der Lautsprecherschall aus ganz anderen Richtungen als der direkte Schall kommt. Im Bereich von 5 ··· 25 ms darf die Erhöhung bis zu 10 dB betragen. Dieser Effekt ist so frappierend,

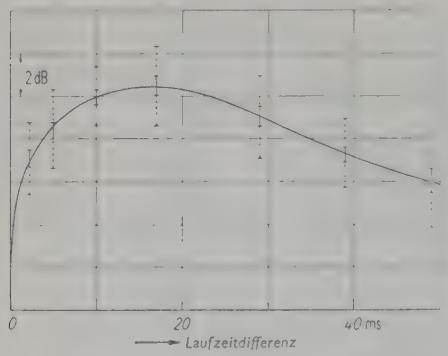


Abb. 6. Verhältnis der Lautsprecherintensität zur Intensität des Primärschalles bei subjektiver Lautstärkegleichheit bzw. bei Mitteneindruck in Abhängigkeit von der Laufzeitdifferenz.

wenn man ihn zum ersten Mal hört, daß es erstaunlich ist, daß man ihn nicht schon sehr viel früher ausgenutzt hat. Auf welcher physiologisch-akustischen oder psycho-akustischen Grundlage er beruht, ist noch nicht geklärt. Übrigens ist das Vorhandensein eines solchen Effektes beim normalen Hören in Räumen notwendig; wäre er nämlich nicht da, würde man infolge der vielen Wandreflexionen nicht mehr die Schallquelle lokalisieren können; (das gleiche trifft natürlich auch für das Hören im Freien zu, da ja in der Regel stets irgendwelche reflektierenden Gegenstände vorhanden sind).

Die zweite Ursache, auf die die wesentliche Verbesserung der Übertragungsanlagen in den letzten Jahren beruht, ist die Möglichkeit, in einfacher Weise gerichtete Lautsprecheranlagen aufzustellen, nämlich in Form einer Zusammenschaltung mehrerer Lautsprecher zu einer Lautsprecherzeile. Damit kann man vorzugsweise die Flächen eines Raumes bestrahlen, auf denen das Publikum sich befindet und wird dadurch von dem Raumnachhall freier, indem man den direkten Schall gegenüber dem Nachhallschall hervorhebt. Ein besonders glücklicher Umstand ist dabei noch, daß man zur Erzielung der Laufzeitverzögerung die gerichteten Lautsprecher vom Publikum aus gesehen hinter den Redner setzen kann, ohne die Rückkopplungsgefahr zu erhöhen.

Lautsprecherzeilen werden auch bei großen Anlagen im Freien benutzt. Man stellt sie auf sehr hohen Türmen auf und richtet sie auf die entfernter liegenden Teile der Zuhörerschaft. Infolge der Bündelung der Schallenergie hat man dann den Vorteil, daß die Abnahme der Schallenergie entlang der Publikumsfläche mit weniger als $1/r$ erfolgt.

Elektroakustische Verzögerungsanlagen mit einem Magnetophon-Laufzeit-System in Räumen sind durch die Versuche bei den Ruhr-Festspielen in Recklinghausen (Schodder, Schröder, Thiele [32]) und vor allem in St. Paul's Cathedral (Parkin [1]) bekannt geworden.

Zum Schluß noch ein Wort über die Möglichkeit, Lautsprecheranlagen auch für künstlerische Zwecke einzusetzen. Die laufzeitrichtige Anlage bewirkt eine erhebliche Schallverstärkung, ohne die Aufmerksamkeit des Zuhörers von der Schallquelle abzuziehen. Wie die eben erwähnten Versuche in Recklinghausen gezeigt haben, hört der durchschnittliche Zuhörer nicht mehr bewußt, ob bei seinem Höreindruck eine Lautsprecheranlage mitwirkt oder nicht. Dies eröffnet ganz neue Perspektiven für die elektroakustischen Anlagen auf künstlerischem Gebiet. Aus Ersparnisgründen werden heute vielfach Mehr-Zwecke-Theater gebaut, d.h. Theater, die man sowohl für Schauspiele, als auch für Opern und sogar für Konzerte benutzt. Raumakustisch ist es ganz unmöglich, einen Raum akustisch optimal für alle drei Kunstgattungen einzurichten. Man kann dann z.B. so vorgehen, daß man den Raum mit einer relativ kurzen Nachhallzeit versieht, die für Schauspiel und Kammermusik ausreichend ist, daß man aber für große Konzert- und Opernaufführungen laufzeitrichtige, möglichst noch über einen Echoraum gehende Anlagen oder sonstige Anordnungen zur Erzeugung künstlichen Nachhalls einsetzt. Derartige Versuche wurden kürzlich mit gutem Erfolg in dem Provisorium der Hamburger Staatsoper durchgeführt. Aber auch die entgegengesetzte Planung ist denkbar, nämlich das Theater für rein musikalische Zwecke herzurichten, also eine verhältnismäßig große Nachhallzeit zuzulassen und für das Schauspiel bzw. für Vorträge Lautsprecheranlagen zur Erhöhung der Sprachverständlichkeit einzusetzen. Man wird gerade in den nächsten Jahren auf diesem Gebiet interessante Entwicklungen erwarten dürfen.

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EXPERIENCES WITH ELECTROACOUSTIC EQUIPMENT USED AT THE OLYMPIC GAMES IN HELSINKI 1952

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Summary. A public address system with loudspeaker arrays as well as a system with loudspeakers spread over a distance, keeping the mutual distance between adjacent units within 17 m, were used. A composite mixture of evenly spoken speech was used for testing the uniformity of sound distribution. Research on echo-effects was carried out using short pulses recorded in the field on level recorders. Both systems mentioned above give uniform distribution and good reproduction, at least out of doors.

Sommaire. On a utilisé un système de "public address" comportant des colonnes de haut-parleurs, ainsi qu'un système avec haut-parleurs disséminés sur une certaine distance, en maintenant un écartement mutuel d'au plus 17 m entre les unités adjacentes. On a contrôlé l'uniformité de distribution du son au moyen d'un mélange composite de phrases prononcées d'une façon égale. On a recherché les effets d'écho au moyen de brèves impulsions reçues sur le terrain par des enregistreurs de niveau sonore. Les deux systèmes ci-dessus produisent une distribution uniforme et donnent une bonne reproduction, du moins à l'extérieur.

Zusammenfassung. Es wurden sowohl Lautsprecherzeilen als auch Systeme mit verteilten Lautsprechern verwendet, wobei der gegenseitige Abstand kleiner als 17 m gehalten wurde. Ein zusammengesetztes Gemisch aus gleichmäßig gesprochener Sprache wurde zur Prüfung der gleichmäßigen Schallverteilung benutzt. Eine Untersuchung auf Echoeffekte wurde mit Hilfe kurzer Impulse, die dann mit Pegelschreibern registriert wurden, durchgeführt. Beide oben erwähnten Lautsprecheranordnungen lieferten, zumindest im Freien, eine gleichmäßige Schallverteilung und eine gute Wiedergabe.

This paper discusses some properties of the sound fields produced by the loudspeaker system used at the Olympic Games in Helsinki 1952.

As a suitable means for testing the uniformity of distribution of sound over an area to be covered, a tape recording was made consisting of a composite mixture of an evenly spoken speech recorded on 12 tape recorders and all these re-recorded simultaneously with a time difference of a few seconds. The final recording, appropriately called the "Monotone", possessed the spectrum of normal speech at a practically unvarying level.

To enable research on echo-effects and sound impulses with too long time differences from different loudspeakers, recorded sound impulses were sent through loudspeakers and registered with level recorders in different places of the area to be covered and thereby making it possible to verify the time differences of the arriving sound impulses and their levels.

Both uninterrupted speech and separately spoken words were of course used for the audibility tests.

Systems with the sound-reproducing units concentrated practically at one point using directed beams of sound (loudspeaker arrays), as well as

systems with a great number of loudspeakers spread over a distance, thereby strictly adhering to the principle of keeping the mutual distance between adjacent units within 17 meters, corresponding to a time difference of 50 milliseconds, were used during the Olympic Games. The considerable experience obtained during the Olympic Games indicates that excellent results can be reached with both systems. So it may be of interest to discuss examples of both loudspeaker systems.

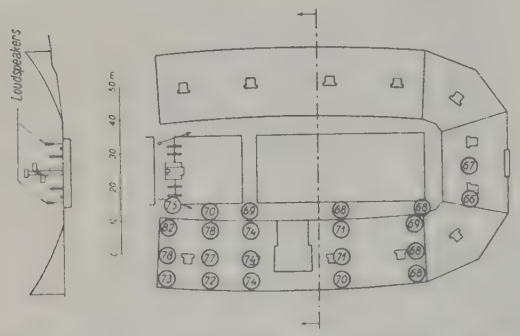


Fig. 1. Directions of the loudspeakers and the distribution of sound (phons) at the Olympic Swimming Pool.

A typical system with the sound source practically at one point only was used at the Swimming Pool (Fig. 1). Two loudspeaker arrays, consisting of six 6 W-loudspeakers each, were set at a distance of 17 meters from each other. Horizontally the sound distribution of the two loudspeaker arrays had an 8-characteristic with the front lobe considerably stronger. The loudspeaker array was 1.8 meters high and the lowest loudspeaker was 3 meters from the ground. Thus the sound was not too loud at ear level, even in front of the loudspeaker. The seats on the stand, close to the loudspeakers and in the same plane, were in an area of zero-radiation of the 8-characteristic. During the preliminary tests echo effects were observed from the restaurant building behind the loudspeakers. They were eliminated by raising the loudspeaker arrays about 1.5 meters, thereby directing the reflection to the ground, closer to the building, and then upwards, avoiding disturbances in the planned audibility area. A very disturbing echo from the score board tower and the opposite stand was prevented, partly by making some partitions penetrable to sound and partly by building them slanting and using perforated letters (Fig. 2).

Fig. 1 shows the sound levels in phons measured in the above-mentioned way at ear level. No echo effects of any importance have been noticed during the tests or by observations made during the Olympic Games or afterwards. It is improbable that such an even sound distribution combined with

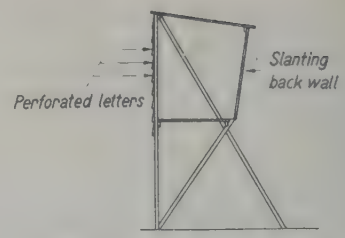


Fig. 2. Score board of the Swimming Pool, showing the perforated letters and the slanting back wall.

good musical quality could have been obtained by using loudspeaker horns.

A typical system with a large number of loudspeakers was used at the Equestrian Stadium (Figs. 3 and 4). Strictly adhering to the principle of

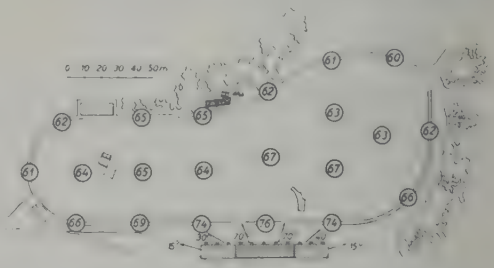


Fig. 3. Directions of the field loudspeakers and the distribution of sound (phons) at the Olympic Equestrian Stadium.

keeping the distance between adjacent loudspeakers within 17 m corresponding to a time difference of 50 ms, it was to be expected that the sound impulses from different loudspeakers would not be distin-

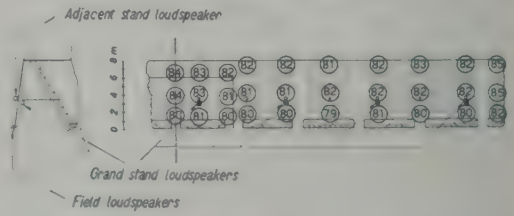


Fig. 4. Distribution of sound (phons) in the grandstand of the Equestrian Stadium.

guished from each other in a disturbing way *). The loudspeakers are in two groups, one for giving information to the grandstand (8 loudspeakers of 6 W) and the adjacent stands (2 loudspeakers of 15 W), and one group for the equestrian field (4 loudspeakers of 15 W). The two groups could of course

*) During the Olympic Winter Games in Oslo this system was used as well at the main Stadium as at the Ice-hockey Stadium, which both were oval. The listener had the feeling of being in a closed space, because the successive loudspeaker impulses produced an effect giving the impression of reverberation. The clearness of speech was perfect.

be used simultaneously. It was feared that the sound reflections from the opposite terrain, the hill and the wood would cause difficulties. Fig. 3 shows the distribution of sound in the area in question and Fig. 4 in the grandstand. Echo recordings were made in the worst affected parts of the whole area, in the grandstand. Fig. 5 shows an impulse and the

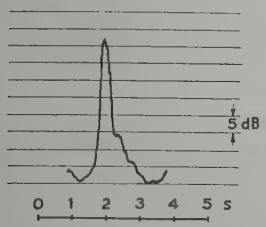


Fig. 5. Echo-effect in the Equestrian Stadium.

corresponding echo arriving 480 ms later and being 28 dB down, as registered with a high-speed level recorder. Here as well the intelligibility was perfect.

It was very difficult to obtain good audibility during the fencing competitions taking place on two courts in a tennis hall with an arched ceiling and with a volume of 16 000 m³, Fig. 6. The reverberation time in this hall was about 4.4 s and the flutter-echos were extremely bad. The conclusion was made that the only way to avoid the disadvantages of too long a reverberation was a correctly

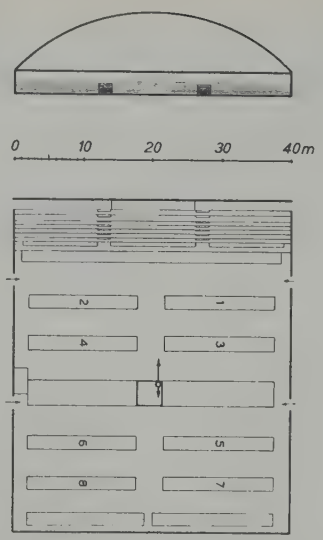


Fig. 6. Tennis hall used for the fencing competitions.

beamed sound distribution. The loudspeaker arrays placed as in the plan gave a good intelligibility in the whole area.

Loudspeakers with time-delayed feeding were not used during the Olympic Games, because the systems were used for music and information with invisible speakers in contrast to artists performing for an audience.

FORTSCHRITTE IN DER ENTWICKLUNG VON LAUTSPRECHERN

von G. Buchmann

Standard Laboratorium der C. Lorenz AG., Stuttgart

Zusammenfassung. Untersuchungen zur Verbesserung der in Rundfunkgeräten verwendeten elektrodynamischen Lautsprecher führten zu einer Membran mit etwa parabelförmigen Mantellinien. Eine zusätzliche Randdämpfung glättet den Verlauf der Frequenzkurven und vermindert die Untertonbildung. Der Schwingspulenstrom übt eine Rückwirkung auf das Magnetfeld aus, die nichtlineare Verzerrungen verursacht. Die in der Membran auftretenden Kräfte können eine Amplitudenmodulation hervorrufen.

Summary. Attemps to improve electrodynamic loudspeakers used for broadcasting lead to a membrane with quasi-parabolic trace. Damping applied to the edges smooths out the frequency responses and reduces the subharmonic content. The current in the coil reacts back on the magnetic field which produces non-linear distortion. The forces set up in the membrane can cause amplitude modulation.

Sommaire. Les recherches faites en vue de perfectionner les haut-parleurs électrodynamiques employés dans les installations radio ont conduit à imaginer une membrane de génératrice sensiblement parabolique. Un amortissement supplémentaire au bord uniformise la courbe de réponse en fréquence et réduit la formation de vibrations sous-harmoniques. Le courant dans la bobine exerce une réaction sur le champ magnétique, ce qui donne lieu à des distorsions non-linéaires. Les forces prenant naissance dans la membrane peuvent produire une modulation en amplitude.

Die Lautsprecherentwicklung hat heute einen Stand erreicht, bei dem weitere Verbesserungen der Wiedergabequalität nur noch in kleinen Schritten erfolgen. Bei entsprechendem Aufwand ist ein so hoher Grad an Originaltreue möglich, daß die noch hörbaren Unterschiede nicht mehr aus den Übertragungseigenschaften der Lautsprecher allein erklärt werden können.

Die Höhe des Aufwandes spielt aber bei Rundfunkempfängern eine entscheidende Rolle. Hier soll möglichst ein einziger Lautsprecher zu einem Herstellungspreise, der nur wenig über dem Einkaufspreis des Magneten liegt, ein Optimum an Klangqualität erreichen. Dazu kommt noch, daß die Anforderungen an diese Lautsprecher durch Einführung des UKW-FM-Rundfunks erheblich gestiegen sind. Während im Mittelwellenbereich das elektrisch übertragene Band im Höchstfalle bis 6000 Hz reicht, sollen im UKW-Bereich auch Frequenzen über 10 000 Hz noch wiedergegeben werden. Eine gewisse Erleichterung der Forderungen ist dadurch gegeben, daß man den Frequenzgang des Lautsprechers, der im allgemeinen ein unlösbarer Bestandteil des Empfängers ist, elektrisch korrigieren kann. Der Umfang dieser Korrektur ist jedoch begrenzt. Steile Sprünge und tiefe Einbrüche entziehen sich dem elektrischen Ausgleich ebenso wie nichtlineare Verzerrungen. Die Rundfunklautsprecher sollten deshalb vor allem einen möglichst glatten Verlauf der Frequenzkurve aufweisen. Der Abfall an den Grenzen des Übertragungsbereiches soll nicht zu plötzlich erfolgen, da dies störende Einschwingvorgänge verursachen würde [1]. Eine gewisse Betonung der tiefen Töne ist wegen der meist kleinen Gehäuse erwünscht.

Die Wirkungsweise des dynamischen Lautsprechers ist allgemein bekannt. Wenn man den Magneten, dessen magnetische Energie im Luftspalt in einem nahezu konstanten Verhältnis zum Preise steht, als gegeben ansieht, so liegt auch die optimale Dimensionierung der Tauchspule fest. Der Frequenzgang und der Wirkungsgrad des Lautsprechers hängen dann nur noch von der Ausführung der Membran ab.

Außer für Spezialzwecke ist die geschöpfte Papiermembran noch unübertroffen. Mit der Stoffzusammensetzung lassen sich Steife und innere Reibung in weiten Grenzen beeinflussen. Dagegen hat sich eine Abwandlung der bisher üblichen Form als günstig erwiesen. Die neue Form ist flacher, ihre Krümmung ist in der Mitte sehr stark und nimmt zum Rande hin ab; die Mantellinien haben etwa Parabelform (Abb. 1). Dies hat zur Folge, daß die

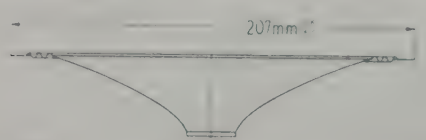


Abb. 1. Membran eines Lautsprechers mit 215 mm Korbdurchmesser.

Frequenzkurve nicht den steilen Anstieg im Bereich der höheren Töne zeigt, wie dies besonders ausgeprägt bei Lautsprechern mit geradem Konus zu beobachten ist. Dafür ist aber der Bereich der wirksamen Abstrahlung zu den hohen Frequenzen hin weiter ausgedehnt und die Richtwirkung weniger ausgeprägt.

Von besonderer Bedeutung für die Glättung der Frequenzkurve ist eine ausreichende Dämpfung der Membraneigenschwingungen, zu denen auch die ausgeprägte Resonanz der Membraneinspannung gehört (Abb. 2, Kurve 1). Verwendet man zu diesem Zweck Papier mit hoher innerer Reibung, so wird gleichzeitig die Abstrahlung der hohen Töne verschlechtert. Es ist deshalb vorteilhafter, die Membran aus hartem dämpfungsärmeren Stoff herzustellen und dafür eine zusätzliche Dämpfung am Rande anzubringen. Dies kann z.B. durch einen Filzbelag geschehen. Am besten verwendet man einen An-

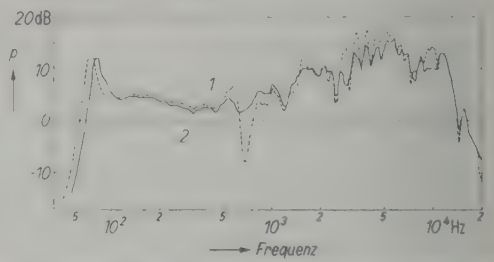


Abb. 2. Einfluß einer Randdämpfung auf den Frequenzgang des Schalldruckes. Frequenzkurven ohne (1) bzw. mit Dämpfung (2).

strich, wie er in ähnlicher Form für das Entdröhnen von Autokarosserien benutzt wird. Das Anstrichmittel behält plastisch-elastische Eigenschaften und soll beim Auftragen möglichst nur oberflächlich am Papier haften und nicht eindringen. Es verklebt sonst die Papierfasern untereinander und macht die Einspannung hart.

Das Resultat dieser Behandlung (Abb. 2, Kurve 2) ist bei richtiger, durch Versuch zu ermittelnder Bemessung, daß die von der Schwingspule ausgehenden Biegewellen am Rande stark geschwächt reflektiert und die Amplituden der stehenden Wellen wesentlich herabgesetzt werden. Auch die Suboktavschwingungen (son rauque) werden wesentlich vermindert. Der Wirkungsgrad des Lautsprechers wird durch diese Behandlung nicht merklich verschlechtert, da der Belag am Rande aufgetragen zur schwingenden Masse wenig beiträgt.

Mit zunehmender Frequenz wird die Wirkung der Randdämpfung geringer. Man kann auch die höheren Resonanzen noch dämpfen, wenn man auch die Membranfläche selbst bestreicht. Man muß dann aber eine Verminderung des Wirkungsgrades durch die Zusatzmasse in Kauf nehmen.

Die Baßresonanz des Lautsprechers macht sich häufig durch störende Einschwingvorgänge bemerkbar, besonders bei schwachem Magnetfeld oder

hohem Innenwiderstand der Spannungsquelle. Man verwendet heute poröse Gewebezentriermembranen, deren geeignet abgestimmter Strömungswiderstand die nötige Dämpfung liefert, und die außerdem den Magnetspalt vor Verschmutzung schützen.

Für die Entstehung nichtlinearer Verzerrungen im Lautsprecher gibt es eine ganze Reihe möglicher Ursachen. Bekannt sind die von der Membranamplitude abhängigen Verzerrungen, hervorgerufen durch:

Nichtlinearität der Einspannungsfederung,
Dopplereffekt,
inhomogenes Magnetfeld,
Amplitudenabhängigkeit des Scheinwiderstandes.

Auf den Mechanismus aller Einflüsse kann hier nicht eingegangen werden. Es genügt, ihre Auswirkungen kurz zu skizzieren.

Die nichtlineare Einspannung verursacht harmonische Obertöne im Bereich der tiefsten Frequenzen. Mit wachsender Frequenz ist das Schwingungssystem in zunehmendem Maße massengehemmt; der Einfluß der Einspannung verschwindet. Durch eine ausreichende Anzahl von Randrillen kann die Nichtlinearität vernachlässigbar klein gehalten werden (Abb. 1). Die erste Rille an der Randeinspannung soll ebenso wie die am Beginn des Konus erhaben ausgeführt werden. Beginnt die Wellung am Rande mit einer Vertiefung, so wird die Federkennlinie unsymmetrisch und die Membranen neigen dazu, sich nach einer Seite zu verlagern. Die Messung der Federeigenschaften der in Abb. 1 gezeigten Membran ergibt bei einer Amplitude von 2 mm einen kubischen Klirrfaktor von 1,5%. Diese Amplitude entspricht bei 100 Hz einer abgestrahlten Schallleistung von ca. 0,3 W.

Der Dopplereffekt verursacht eine Frequenzmodulation, hervorgerufen durch Töne, welche im Frequenzbereich des Anstiegs des Strahlungswiderstandes liegen. Auf ihn hat der Konstrukteur bei gegebenen Membranabmessungen nur insofern Einfluß, als er durch günstige Formung des Gehäuses den Strahlungswiderstand bei niedrigen Frequenzen groß zu machen versucht.

Das inhomogene Magnetfeld verursacht harmonische Obertöne und Amplitudenmodulationen. Das gleiche gilt für den veränderlichen Scheinwiderstand. Diese Einflüsse können nur durch Vergrößerung des Magneten (tieferer Spalt) oder Einbuße an Wirkungsgrad (längere Tauchspule) vermindert werden.

Alle eben genannten Verzerrungen sind an das Vorhandensein großer Amplituden, d.h. tiefer Frequenzen gebunden. Dagegen können die vom Schwingspulenstrom abhängigen Verzerrungen bei allen Frequenzen auftreten. Kaum beachtet wurden bisher die Rückwirkung des Schwingspulenstromes auf das Magnetfeld und die Änderung der Strahlungseigenschaften infolge von Zug- und Schubspannungen in der Membran.

Durch die Unsymmetrie des Magnetaufbaus gegenüber der Spule erzeugt der Schwingspulenstrom I eine zusätzliche Induktion, die angenähert $\frac{1}{2}\,\mu_0\, w\, I/\delta$ ist, wenn w die Windungszahl der Spule und δ die Spaltweite sind. Auf den Spulendraht von der Länge l wird eine Kraft ausgeübt

$$P' = \tfrac{1}{2}\,\mu_0\,\frac{w\,l}{\delta}\,I^2.$$

Sie wächst quadratisch mit dem Strom und überlagert sich der Kraft $P = BIl$ (B = Induktion im Spalt). Es entsteht ein quadratischer Klirrfaktor $k_2 = \frac{1}{2}\,P'/P = \frac{1}{4}\,\mu_0\, w\, I/\delta B$. Bei einem mittleren Lautsprecher beträgt dieser Klirrfaktor etwa 1 ··· 2% bei voller Leistung. Große Spaltweite und hohe Magnetinduktion sind in diesem Falle anzustreben.

Daß auch die von der Tauchspule auf die Membran ausgeübten Kräfte die akustischen Membraneigenschaften verändern können, geht aus Abb. 3

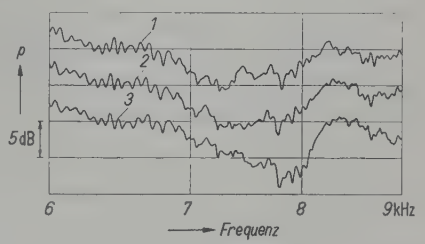


Abb. 3. Frequenzgang des Schalldruckes in Abhängigkeit von einem durch die Tauchspule fließenden Gleichstrom;
Kurve 1: Gleichstrom 1 A,
Kurve 2: ohne Gleichstrom,
Kurve 3: Gleichstrom umgepolt 1 A.

hervor. Sie zeigt einen Ausschnitt aus der Frequenzkurve eines Lautsprechers mit einem durch die Spule fließenden Gleichstrom als Parameter. Der besseren Übersicht wegen sind die Kurven um 5 dB gegeneinander versetzt. Man erkennt, daß ihr Verlauf verschieden ist. Die Abweichungen betragen in diesem besonders ungünstigen Fall max. 4 dB. Der angewendete Gleichstrom entspricht einer elektrischen NF-Leistung von 2 VA. Dieses Verhalten der Membran verursacht eine Amplitudenmodulation. Im Gegensatz zu der amplitudenabhängigen Nichtlinearität kommen bei den stromabhängigen Verzerrungen auch Amplitudenmodulationen mit hoher Modulationsfrequenz zustande. Für diese ist das Ohr, wie Untersuchungen gezeigt haben [2], besonders empfindlich. Sie müssen deshalb stärker als bisher beachtet werden.

Die Abb. 4 stellt die Frequenzkurven und den Klirrfaktor eines nach den besprochenen Richtlinien

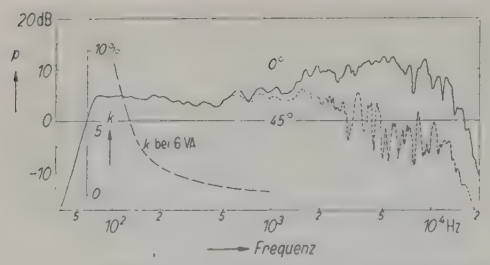


Abb. 4. Frequenzgang von Schalldruck und Klirrfaktor eines 6 W-Lautsprechers. Lautsprecher betrieben mit 50 mVA bei 1000 Hz; $Z_i = Z_a$; Meßabstand 1,25 m; 0 dB ≙ 0,1 N/m². Aufnahme des Klirrfaktors bei 6 VA elektrischer Scheinleistung.

konstruierten 6 W-Lautsprechers dar. Die durch Wechselbeanspruchung der Membran hervorgerufene Amplitudenmodulation beträgt bei 4 W elektrischer Leistung max. 5%.

Schrifttum

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[2] Zwicker, E., Grenzen der Hörbarkeit der AM und FM eines Tones. Acustica 2 [1952], Beiheft 2, AB 125.
Ebel, H., Das Hören von Amplitudenmodulationen. Acustica 2 [1952], Beiheft 4, AB 246, u.a.

Diskussion

Smit, H. J. A.: Hohe Sättigung des Eisenschlusses im Magneten müßte die Rückwirkung herabsetzen.

Vortragender: Das trifft zu. Die Sättigung ist jedoch für den Magneten unwirtschaftlich.

Holle, W.: Kann eine plausible Erklärung dafür gegeben werden, daß sich eine günstigere Federkennlinie ergibt, wenn die Randrille mit einer Erhebung beginnt?

Vortragender: Nein. Diese Feststellung ist lediglich durch Messungen immer wieder bestätigt worden.

Gayford, M. L.: Sind Messungen der Einschwingvorgänge an diesen Lautsprechern vorgenommen worden?

Vortragender: Wir haben die Einschwingvorgänge nur bei einigen Frequenzen geprüft. Sie sind, was aus der starken Dämpfung einleuchtet, tatsächlich kürzer als bei üblichen Lautsprechern.

Kösters, H.: Die Schalldruckkurve ist bei der flachen Membran besser. Ich befürchte, daß deshalb aber die Frequenzkurve der Leistung schlechter ist.

Vortragender: Die Einebnung der Schalldruckkurve erfolgt bei der flachen Membran zum großen Teil durch Verringerung der Richtwirkung, so daß die nach den Seiten abgestrahlte Leistung zunimmt.

Skudrzyk, E.: Bei Untersuchungen des Einflußes der Membranform zeigte sich, daß die Lage der radialen Obertöne der Membran sehr wichtig ist. Die Grundresonanz übernimmt die Schallabstrahlung bis etwa 1000 Hz (übliche Lautsprecher). Dann aber bleibt der Strahlungswiderstand konstant, die Schallabstrahlung würde abfallen, wenn nicht die Membran unterteilt schwingen, d.h. die Obertöne einsetzen würden. Die alten Nawi-Membranen sind offenbar zu steif, ihre Obertöne liegen zu hoch, so daß die Frequenzkurve oft Löcher im Bereich von 1,5 ··· 3 kHz aufweist. Die hier erläuterte flachere Membran stellt offenbar die Lösung dieses Problems dar.

Vortragender: Die Eigenfrequenzen dieser Membran sind tatsächlich zu den tiefen Frequenzen hin verschoben. Sie würden ohne Randdämpfung auch stark in Erscheinung treten.

LES CONDITIONS DE VÉRIFICATION ACOUSTIQUE DES HAUT-PARLEURS

par P. Chavasse et R. Lehmann

Centre National d'Études des Télécommunications, Paris

Sommaire. Les mesures acoustiques à effectuer sur les haut-parleurs sont délicates et exigent un appareillage spécial et de haute qualité, ainsi qu'un laboratoire spécialisé. Souvent, les mesures sont faites dans des conditions très différentes quand on passe d'un laboratoire à un autre et les résultats obtenus sont difficilement comparables. C'est dans le but de normaliser les divers essais auxquels doivent être soumis les haut-parleurs qu'est présenté le travail suivant qui passe successivement en revue: la mesure de la caractéristique de fréquence en régime permanent; la mesure de la directivité; la mesure de la distorsion harmonique; la mesure de l'intermodulation; la mesure de la caractéristique d'impédance; la mesure du rendement; la mesure de la courbe de réponse en régime transitoire.

Zusammenfassung. Messungen mit Lautsprechern werden oft in verschiedenen Laboratorien unter sehr abweichenden Bedingungen durchgeführt, so daß die Resultate schwer vergleichbar sind. Um die verschiedenen Prüfungen, denen man die Lautsprecher unterwerfen muß, zu normalisieren, werden hier die Messungen der folgenden Größen betrachtet: Frequenzkurve im stationaren Zustand, Richtcharakteristik, harmonische Verzerrungen, Kombinationstöne, Impedanzkurve, Wirkungsgrad, Ausgleichsvorgänge.

Summary. Measurements made on loudspeakers have to be delicate and demand high-quality apparatus in a special laboratory. With the object of standardising measurements in different laboratories, often difficult to compare, the procedure to be adopted for the following measurements is laid down, viz., frequency characteristic in steady state, directivity, harmonic distortion, intermodulation, impedance characteristic, output, transient response.

1. Généralités

Les méthodes de mesure relatives aux haut-parleurs sont difficiles et délicates à mettre en oeuvre, et, par défaut d'une doctrine définie, sont très diversement exécutées dans les divers laboratoires; cette situation apparaît un peu paradoxale quand on la rapproche de celle où l'on se trouve dans le domaine de la transmission radioélectrique; cela tient sans doute à la méconnaissance des facteurs acoustiques de la part de beaucoup de techniciens et, peut être aussi, à la complexité théorique et pratique des problèmes que pose l'acoustique. Il semble même que le haut-parleur, quoiqu'ayant un rôle primordial dans la reproduction des sons et dans la qualité globale des transmissions radiotéléphoniques, soit souvent traité "comme le parent pauvre" de la chaîne électroacoustique, qui commence au microphone du poste émetteur pour se terminer à l'oreille de l'usager. Cette situation est en partie vraisemblablement due au fait que les vérifications qu'il doit subir demandent un matériel, des techniciens et des installations très spécialisés et, d'autre part, en raison de la priorité accordée au relevé de la courbe de réponse en fonction de la fréquence, au détriment des notions de distorsion, d'intermodulation et de réponse en régime transitoire, entre autres, qui sont presque toujours systématiquement négligées.

La présentation des catalogues des divers constructeurs met très nettement en évidence cet état de chose; de plus, une incertitude règne souvent sur les conditions dans lesquelles ont été effectuées les mesures dont les résultats sont présentés. C'est en vue des discussions techniques engagées pour parvenir à un accord international relatif, d'une part, à la nature des essais à effectuer et, d'autre part, aux méthodes de mesures, qu'a été préparé ce travail dans lequel ont été volontairement laissées de côté toutes les questions concernant l'appareillage.

À la connaissance des auteurs, avant la deuxième guerre mondiale, divers pays avaient créé des commissions chargées de l'étude de ces questions, afin d'élaborer des normes nationales, et avec l'espoir de pouvoir, dans la suite discuter la mise au point d'un accord international. La guerre a interrompu ces divers travaux, du moins en Europe; nous croyons savoir, en effet, que seuls les États-Unis [1] ont actuellement formulé des recommandations relatives aux conditions de vérification acoustique des haut-parleurs. Cette norme américaine préconise l'exécution des essais relatifs aux cinq points suivants:

courbe de réponse en régime permanent,
distorsion harmonique,
directivité,
rendement,
caractéristique d'impédance.

Le comité acoustique allemand, avant la guerre, préconisait déjà les 4 premiers des essais précédents, ainsi qu'une mesure de la charge maximum tolérable. Nous pensons, de notre côté, qu'il est nécessaire de conserver tous les essais proposés dans la norme américaine, mais d'y ajouter, dans la mesure du possible, un relevé de la caractéristique de réponse en régime transitoire. En ce qui concerne la charge maximum, elle semble encore très difficile à définir, car elle dépend, dans de très grandes proportions, de la fréquence; aussi proposons-nous d'en remplacer la spécification par celle de la puissance nominale du haut-parleur, que devrait garantir le constructeur et qui pourrait concerner la puissance électrique dissipée entre ses bornes d'entrée à la fréquence 1000 Hz, pour laquelle la distorsion harmonique mesurée à 2 m dans l'axe de haut-parleur serait de 3%.

2. La mesure des caractéristiques acoustiques des haut-parleurs

A notre avis, toutes les caractéristiques doivent être mesurées en champ acoustique libre ou, tout au moins, dans des conditions théoriques équivalentes, c'est-à-dire dans une salle muette. De plus, le mode de fixation du haut-parleur doit être spécifié avec précision et, à cet effet, nous proposons:

a) pour les haut-parleurs à membrane, il serait désirable de les fixer sur un écran (baffle) infini, qui serait ainsi précisé, car la forme et la dimension de l'écran exercent une influence très appréciable sur la forme de la caractéristique de fréquence, ainsi que le montrent les figures 1 et 2. Comme il est souvent difficile d'opérer avec un écran infini, nous pensons qu'un écran carré de 2 m de côté, avec fixation du haut-parleur en son centre, donnerait généralement

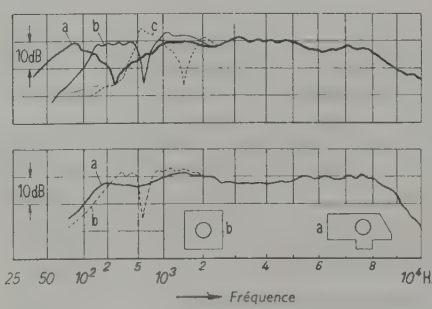


Fig. 1. Caractéristiques de fréquences d'un haut-parleur à membrane fixé sur des écrans carrés de dimensions variées:
a) écran carré de 160 cm de côté,
b) „ „ „ 80 „ „ „ ,
c) „ „ „ 40 „ „ „ .

Fig. 2. Caractéristiques de fréquences d'un haut-parleur à membrane fixé sur des écrans de formes diverses.

toute satisfaction, quoique certains auteurs préconisent des écrans irréguliers [2].

b) pour les haut-parleurs à chambre de compression, il est indispensable d'exécuter les mesures en les munissant des pavillons avec lesquels ils doivent fonctionner.

Enfin, les positions respectives du microphone de mesure et du haut-parleur à étudier sont également à préciser. Nous proposerons de placer le microphone dans l'axe du haut-parleur, à une distance de 2 m de celui-ci (distance de 2 m comptée à partir de l'écran du haut-parleur pour les appareils à membranes et à partir du plan de base du pavillon pour les appareils fonctionnant avec pavillon).

a. *Caractéristique de fréquence*

Il serait désirable d'effectuer ce relevé en alimentant le haut-parleur avec une puissance électrique constante, ce qui n'est malheureusement pas possible à cause de la variation de l'impédance en fonction de la fréquence, lorsque l'on trace la caractéristique de facon continue. Pratiquement, il est recommandé d'exécuter ce relevé en maintenant la tension constante aux bornes du haut-parleur. Deux méthodes automatiques sont actuellement utilisées à cet effet:

α) méthode basée sur l'emploi de fréquences pures discrètes,

β) méthode basée sur l'emploi d'un bruit complexe.

La première méthode est préférable, car elle est plus précise. L'enregistrement doit être exécuté, pour la bande 25 ··· 15 000 Hz, parcourue en un temps de 3 minutes au moins, dans les conditions précisées dans le paragraphe précédent. Nous préconisons d'exécuter trois relevés pour trois tensions différentes, l'une étant celle correspondant à la puissance nominale du haut-parleur, les deux autres étant respectivement égales à la moitié et au double de cette tension. Les résultats devront être présentés en portant les fréquences en abscisses suivant une échelle logarithmique et l'efficacité en ordonnées, suivant une échelle linéaire.

La deuxième méthode, indiquée par FREYGANG en 1938 [3] et reprise en 1942 par OLNEY [4], nécessite l'utilisation d'un analyseur de fréquences, mais a, par contre, l'avantage de permettre l'étalonnage dans une salle quelconque.

b. *Caractéristiques de directivité*

Nous préconisons d'exécuter les courbes de directivité pour les fréquences 125, 400, 1000 et 5000 Hz, valeurs d'ailleurs non limitatives. Nous pensons également que l'enregistrement continu est préférable, ce qui signifie que l'on doit enregistrer la variation de la pression sonore en fonction de l'angle d'une manière automatique. Il est souhaitable de présenter les résultats sous la forme d'un diagramme polaire.

c. *Distorsion harmonique et intermodulation.*

Ces deux caractéristiques sont très importantes et ne font pas double emploi, mais se complètent utilement; toutes deux doivent être mesurées à l'aide d'un analyseur de fréquences, le bruit de fond ne devant pas être compris dans le taux de distorsion harmonique.

α) distorsion harmonique: nous préconisons d'exécuter les mesures aux fréquences 125, 400 et 1000 Hz, pour les haut-parleurs à pavillon; à la fréquence de résonance de la membrane et aux fréquences 400 et 1000 Hz pour les haut-parleurs à membrane. Les tensions pour lesquelles ces mesures doivent être faites seront celles pour lesquelles auront été relevées les courbes de réponse. Il faut naturellement, lors de ces essais, veiller à ce que la tension électrique aux bornes du haut-parleur soit aussi pure que possible, ce qui n'est pas toujours facile à satisfaire.

β) intermodulation: les mesures d'intermodulation ont dû être faites, à l'origine, sur des microphones à charbon [5] et étendues aux haut-parleurs en 1941 [6]. La difficulté réside dans le choix des groupes de fréquences à utiliser et pour lesquels les avis sont encore partagés. Notre expérience nous conduit à proposer les 4 groupes suivants:

1) F_1 = fréquence de résonance de la membrane (ou le double de la fréquence de coupure de pavillon pour les haut-parleurs à pavillon),
   F_2 = 1000 Hz,
2) F_1 = idem à 1),
   F_2 = 400 Hz,
3) F_1 = 100 Hz,
   F_2 = 1000 Hz,
4) F_1 = 400 Hz,
   F_2 = 7000 Hz.

Dans les essais 1 et 2, les tensions relatives aux fréquences F_1 et F_2 seront égales, tandis qu'elles seront dans le rapport 4 pour les essais 3 et 4. Nous proposons le calcul du taux d'intermodulation par la formule suivante:

$$T(\%) = 100 \times \sqrt{\frac{P^2_{2-1} + P^2_{2+1} + P^2_{(2\times2)-1} + P^2_{(2\times2)+1}}{P^2_1 + P^2_2}}.$$

d. *Caractéristique d'impédance*

Lors de l'exécution de cette mesure, le haut-parleur doit normalement rayonner en champ libre et la tension doit être celle qui a été définie pour la production de la puissance nominale. Un simple pont d'impédance peut être utilisé pour mesurer

l'impédance du haut-parleur, mais on peut avantageusement employer une méthode simple signalée par VENEKLASEN [7]. Lorsque l'on constate une variation de l'impédance en fonction de la puissance électrique fournie au haut-parleur, il est nécessaire d'exécuter la mesure pour plusieurs valeurs de la tension d'alimentation.

e. *Caractéristique de rendement*

Si la mesure de la puissance électrique fournie au haut-parleur n'offre pas de difficulté, il n'en est pas de même de la mesure de la puissance acoustique totale rayonnée. Plusieurs méthodes ont été proposées pour la mesure de cette dernière:

α) méthode de la chambre réverbérante de MEYER et JUST,

β) méthode du champ acoustique libre,

γ) méthode de l'impédance motionnelle de KENNELLY et PIERCE.

Il nous paraît difficile de préconiser l'une de ces méthodes de préférence aux autres, car, comme le montre la figure 3, elles donnent des résultats très comparables.

Cependant, la deuxième méthode est plus rigou-

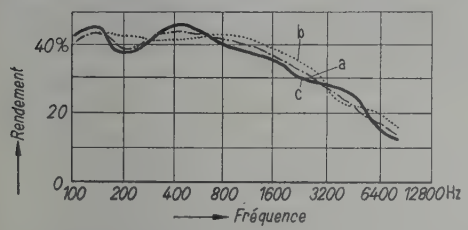


Fig. 3. Caractéristiques de rendement d'un haut-parleur à pavillon, déterminées par trois méthodes différentes.
a) méthode de l'impédance motionnelle,
b) méthode du champ acoustique libre,
c) méthode de la chambre réverbérante.

reuse, quoique plus délicate, mais elle ne semble pas donner prise à critique si les mesures sont minutieusement conduites. Nous préconisons l'exécution des mesures pour les fréquences 125, 400, 1000 et 5000 Hz, sous les conditions de tension déjà définies dans les paragraphes précédents.

Certains auteurs utilisant la troisième méthode exécutent des mesures d'impédance en plaçant successivement le haut-parleur dans l'air et dans le vide; dans ce dernier cas, la charge du diaphragme n'est pas normale et les pertes ne sont pas les mêmes qu'en utilisation courante, de telle sorte que les calculs de rendement risquent d'être entachés d'erreurs à la base; par contre, la mesure est très simple.

f. *Caractéristique en régime transitoire*

C'est une mesure primordiale mais très délicate à exécuter. Les avis sont partagés sur le choix des ondes à utiliser: ondes carrées ou trains d'ondes sinusoïdales ou, enfin, impulsions (dites de DIRAC). Nous pensons que les deux dernières méthodes correspondent avec plus de rigueur aux conditions d'utilisation pratique des haut-parleurs et sommes, en conséquence, partisans de prendre des signaux électriques impulsionnels [8] dont la forme est donnée dans la figure 4a et de mesurer le temps de décroissance δt du son (voir Fig. 4b); la fréquence

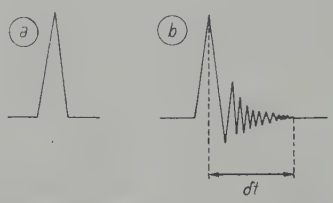


Fig. 4. Les signaux utilisés en régime transitoire:
a) forme approximative du signal impulsionnel appliqué sur la bobine mobile du haut-parleur 10 à 15 fois par seconde,
b) forme du signal fourni par un microphone placé devant le haut-parleur.

de répétition de ces signaux pourrait avantageusement être de l'ordre de 10 à 15 par seconde car les oscillations excitées par des ondes correspondant aux fréquences propres des haut-parleurs semblent toujours amorties au bout d'une quinzaine de périodes environ.

3. Conclusions

C'est avec le souci de donner une base précise aux discussions que ne peut manquer de provoquer l'étude des conditions normales d'utilisation des haut-parleurs que le présent document a été rédigé.

Nous serions heureux qu'il contribuât à susciter des remarques, des critiques et des compléments constructifs.

Nous pensons qu'il est indispensable, dans l'avenir, grâce à un accord universel, de spécifier des méthodes précises et unifiées, afin que constructeurs et laboratoires exécutent des essais analogues, dont les résultats soient rapidement et directement comparables, à l'abri de toute controverse.

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Discussion

SACERDOTE, G. G.: It is interesting to measure the directivity of nonlinear distortion: very often a loudspeaker with little distortion on the axis has great distortion in other directions. Moreover the measurement of sub-harmonics of loudspeakers seems important; this point is often disregarded in ordinary measurements of distortion. A method is proposed to measure the quality of loudspeakers by recording logatomes passing several times through the loudspeakers under test. Good loudspeakers can stand 8 iterations before the intelligibility becomes less than 50%, bad loudspeakers not more than 2 to 3.

CONFÉRENCIER: 1) Il est également intéressant de mesurer la distorsion harmonique pour divers angles. 2) Il est aussi important de rechercher les fréquences sous-harmoniques émises par le haut-parleur. 3) La mesure de la qualité par enregistrement de logatomes est naturellement intéressante, mais n'a de valeur que lorsque l'écoute est faite par plusieurs personnes, ce qui complique l'essai. Par contre, c'est une mesure qui tient compte de toutes les propriétés du haut-parleur. Il faut, de plus, connaître la qualité du microphone et du magnétophone utilisés.

GAVREAU, V.: Le système d'enregistrements successifs sur un ruban magnétique fait intervenir 3 éléments: le haut-parleur étudié, le microphone, le ruban. Et nous avons pu constater hier la qualité de tels enregistrements ... — La mesure de netteté préconisée par M. SACERDOTE peut-elle dans ces conditions donner une idée exacte de la qualité du haut-parleur seul?

CONFÉRENCIER: La mesure préconisée par M. SACERDOTE peut donner des indications très intéressantes si on a étudié, au préalable, la qualité du microphone et du magnétophone utilisés, en particulier leurs distorsions.

FEDERICI, M.: J'aimerais savoir si on peut obtenir avec la méthode de mesure de la puissance d'un haut-parleur par l'impédance motionnelle, une précision suffisante, même avec des haut-parleurs à faible rendement, pour lesquels l'impédance motionnelle n'est qu'une faible partie de l'impédance totale.

CONFÉRENCIER: Si la mesure de l'impédance totale est exécutée avec une grande précision, il est possible d'obtenir une précision suffisante dans la mesure du rendement d'un haut-parleur par la méthode de l'impédance motionnelle. Ainsi, sur des récepteurs téléphoniques, la méthode est également applicable.

BARON, P.: M. CHAVASSE préconise de tracer la courbe de réponse à une distance de 2 m entre 25 et 15000 Hz. Une mesure à 2 m à la fréquence 25 Hz signifie-t-elle quelque chose, et la courbe tracée à 2 m n'est elle pas très différente de celle qu'on obtient aux grandes distances?

CONFÉRENCIER: Pour les basses fréquences, il serait naturellement préférable de tracer la courbe de réponse à une distance plus grande; il faut cependant noter qu'une caractéristique relevée à 2 m du haut-parleur donne déjà des renseignements intéressants et précis.

HÖRBARKEIT NICHTLINEARER VERZERRUNGEN BEI DER ÜBERTRAGUNG VON INSTRUMENTENKLÄNGEN

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Institut für Nachrichtentechnik der Technischen Hochschule, Stuttgart

Zusammenfassung. Obertonarme Klänge sind gegen nichtlineare Verzerrungen viel empfindlicher als obertonreiche Klänge. Die Ursachen und das Maß dieses Unterschiedes werden beschrieben.

Summary. A timbre poor in overtones is much more sensitive to nonlinear distortion than one rich in overtones. The causes and the extent of this difference are discussed.

Sommaire. Les timbres d'instruments pauvres en harmoniques sont beaucoup plus sensibles aux distorsions non-linéaires pendant leur transmission que ceux riches en harmoniques. On indique les raisons de cette différence et son importance.

Bei der Prüfung von hochwertigen Systemen zur Übertragung von Musik beobachtet man, daß die obertonarmen Klänge gewisser Instrumente, z.B. der Flöten, schwerer zu übertragen sind als die obertonreichen Klänge anderer Instrumente, z.B. der Violinen. Auch der aufmerksame Konzertbesucher hört die subjektiven Differenztöne bei den obertonarmen Instrumenten viel leichter als bei den obertonreichen Instrumenten. Auf diesen Unterschied muß man bei der Untersuchung der Hörbarkeit nichtlinearer Verzerrungen also achten.

Es ist lange bekannt, daß die nichtlinearen Verzerrungen am stärksten bei der Übertragung von Zweiklängen stören. Sie machen sich durch die entstehenden Differenztöne und Summentöne bemerkbar. Diese Kombinationstöne werden hörbar, sobald sie die Hörschwelle überschreiten. Das hängt akustisch in erster Linie vom Schalldruck ab, in zweiter Linie von der Frequenz des tiefen Grundtons und vom Intervall, elektrisch von der Krümmung der Kennlinie des Übertragungssystems und von der Zuordnung des Schalldruck-Pegels zum Spannungs-Pegel.

Der Einfluß des *Übertragungssystems* auf die Hörbarkeit der nichtlinearen Verzerrungen ist in der Regel sehr einfach. Man kann annehmen, daß die Stärke der Kombinationsschwingungen, soweit diese auf quadratischen Nichtlinearitäten beruhen, proportional der Amplitude der Spannung ist; soweit diese auf kubischen Nichtlinearitäten beruhen, ist sie proportional dem Quadrat der Spannung. Man kann ferner annehmen, daß bei den gebräuchlichen Übertragungssystemen die Nichtlinearität im mittleren Frequenzgebiet nahezu unabhängig von der Frequenz ist und daß sie nach tiefen Frequenzen zu etwas ansteigt. CCIF läßt bei der größten übertragenen Spannung zwischen 0,1 und 7,5 kHz Klirrfaktoren bis 0,01 zu, unterhalb 0,1 kHz Klirrfaktoren bis 0,03.

Der Einfluß des *Gehörs* auf die Hörbarkeit der nichtlinearen Verzerrungen ist viel verwickelter. Bei tiefen Klängen sind die Summentöne, bei hohen Klängen sind die Differenztöne hörbar, wenn sie die *Hörschwelle* überschritten haben. Für kleine Lautstärken gilt dabei die Hörschwelle, wie man sie mit einem einzelnen Ton mißt. Sie liegt bekanntlich bei tiefen Frequenzen bei ziemlich hohen Schalldrucken, bei mittleren Frequenzen bei tiefen Schalldrucken und steigt bei hohen Frequenzen wieder zu höheren Schalldrucken an. Für große Lautstärken wird die Hörschwelle durch den übertragenen Zweiklang infolge des Verdeckungseffektes verändert. Dabei werden die hohen Summentöne stärker verdeckt als die tiefen Differenztöne. Wir finden deshalb einen Einfluß der Verdeckung auf die Hörbarkeit der nichtlinearen Verzerrungen nur bei tiefen Klängen. Bei großen Lautstärken ändert sich die Hörschwelle stärker als die Amplitude des Schalldruckes. Deshalb nimmt überraschenderweise die Hörbarkeit der Summentöne bei sehr großen Lautstärken wieder ab, wenn man die Verzerrungen konstant hält. Sie werden trotz ihrer großen Schalldrucke von den Originalklängen verdeckt. So bildet sich, immer bei konstant gehaltenen Verzerrungen, für die Hörbarkeit der Summentöne ein günstigster Schalldruck-Pegel heraus, der etwa bei 70 dB liegt.

Schließlich werden diejenigen Kombinationstöne, die auf der Frequenzskala den Originaltönen besonders benachbart sind, stärker verdeckt als die, die weiter ab liegen. Deshalb sind die nichtlinearen Verzerrungen bei engen, hohen Intervallen leichter zu hören als bei weiten oder tiefen Intervallen. Der hierin begründete Einfluß der *Intervallbreite* ist aber nicht erheblich.

Die Grenzen der Hörbarkeit nichtlinearer Verzerrungen gibt man durch den Verlauf der *Klirrschwellen* in der Hörfläche an. Um sie z.B. für eine obertonarme Quint zu messen, hält man im nichtlinearen Teil des Übertragungssystems den Spannungspegel konstant und damit auch die Pegeldifferenzen zwischen Original-Teiltönen und Kombinationstönen. Die Nichtlinearität des Systems kennzeichnet man durch seinen Klirrfaktor. Es ist zweckmäßig, hierfür den Klirrfaktor einer einzelnen Sinus-Spannung zu wählen, deren Effektivwert so groß ist wie der Effektivwert des übertragenen Zweiklanges. Man hält also die Verzerrungen konstant, ändert durch einen Regler vor dem Telephon den Abhörschalldruck und bestimmt die Reglerstellung und damit den Schalldruckpegel, bei dem die gewählten nichtlinearen Verzerrungen eben hörbar werden oder eben verschwinden. Abb. 1 zeigt die

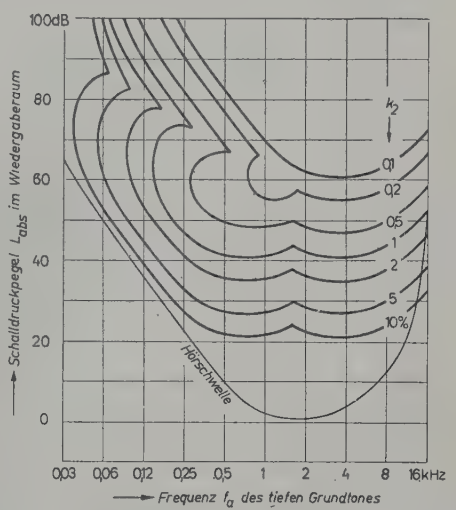


Abb. 1. Klirrschwellen einer quadratisch verzerrten Quint aus Einzelklängen ohne Obertöne.

Klirrschwellen für einen obertonarmen Zweiklang (Quint) bei rein quadratischen Verzerrungen. Die oben beschriebene Wirkung der Verdeckung erkennt man daran, daß bei tiefen Frequenzen jede Klirrschwelle einen kleinsten und einen größten Schalldruck angibt.

Untersucht man obertonreiche Quinten, so findet man einen anderen Verlauf der Klirrschwellen (Abb. 2). Sie liegen etwa 10 dB höher. Das hat verschiedene Gründe: bei gleichem effektiven Pegel haben die einzelnen Teiltöne und damit auch die aus ihnen gebildeten Kombinationstöne eine viel kleinere Amplitude. Liegen die Teiltöne eines Einzelklanges harmonisch, dann liegen auch die von ihnen untereinander gebildeten Kombinationstöne harmonisch und stören kaum. Aber die Kombinationstöne, die durch das Zusammenwirken der Teiltöne beider Einzelklänge entstehen, liegen z.B. bei temperierter Stimmung nichtharmonisch und bedingen die Hörbarkeit der nichtlinearen Verzerrungen. Sie liegen auch relativ eng auf der Frequenzskala, werden nicht mehr einzeln gehört und bilden Geräusche. Ob man diese Geräusche hören kann, hängt davon ab, wie

stark sie sind und wie sehr sie durch die benachbarten Teiltöne des Originalklanges verdeckt werden.

Jede Klirrschwelle gilt nur für feste Aussteuerung des nichtlinearen Systems. In Wirklichkeit wechselt die Aussteuerung je nach der Zuordnung des Schalldruck-Pegels L_a im Aufnahmeraum, des Spannungspegels P_s im Übertragungssystem und des Schalldruck-Pegels L_w im Wiedergaberaum. Ein Beispiel soll das verdeutlichen: ein Kondensatormikrophon habe eine Rauschspannung, die einem Schalldruck-Pegel von 20 dB entspricht. Das Übertragungssystem soll die Anforderungen des CCIF erfüllen: maximaler Spannungspegel um 50 dB über dem Rauschpegel und mit einem Klirrfaktor von 0,01. Läßt man die Rauschpegel von Mikrophon und Übertragungssystem zusammenfallen, so entspricht dem maximalen Spannungspegel ein Schalldruckpegel $L_a = 70$ dB. Nach Abb. 2 ist aber für obertonreiche Klänge ein quadratischer Klirrfaktor von 0,01 nur bei einem Schalldruckpegel $L_w = 52$ dB zulässig. Also muß der Schalldruckpegel im Wiedergaberaum um 18 dB niedriger als im Aufnahmeraum sein. Der Dynamikbereich liegt dann im Wiedergaberaum zwischen 52 dB und 2 dB.

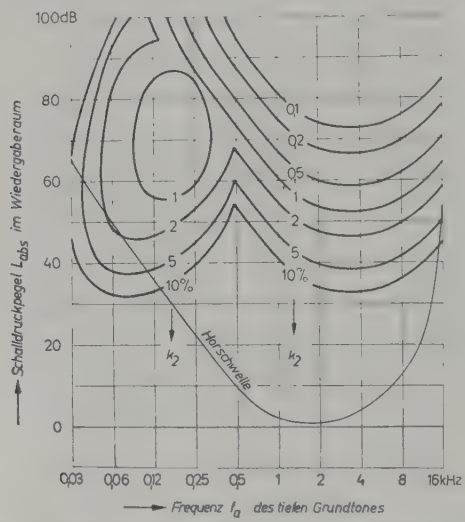


Abb. 2. Klirrschwellen einer quadratisch verzerrten Quint aus Einzelklängen mit je 10 Teiltönen.

Werden nun obertonarme Klänge übertragen, so darf man das Übertragungssystem nicht voll aussteuern, denn bei einem Schalldruckpegel im Wiedergaberaum von 52 dB ist nur ein Klirrfaktor von 0,003 zulässig. Nimmt man an, daß der Klirrfaktor aus einer quadratischen Verzerrung stammt und der Spannung proportional wächst, so findet man aus Abb. 1, daß ein Schalldruckpegel von 46 dB mit einem Klirrfaktor von 0,006 zulässig ist.

Dieses kleine Zahlenbeispiel illustriert die Anwendung der Klirrschwellen bei der Beurteilung eines Übertragungssystems. Es zeigt aber auch, daß man aus den Klirrschwellen des Ohres Schlüsse auf die zulässigen nichtlinearen Verzerrungen des Übertragungssystems erst ziehen kann, wenn man Vereinbarungen über den Dynamikbereich und über die Zuordnung der Schalldrucke im Aufnahmebereich und im Wiedergaberaum geschaffen hat.

Diskussion

LEEUWEN, F. J. van: Wie stark streuen die Klirrschwellen bei verschiedenen Versuchspersonen?

VORTRAGENDER: Entsprechend der individuellen Hörschwelle bei gesundem Gehör etwa $\pm$ 10 dB.

STIER, H.: Sind Untersuchungen mit nichtlinearen Verzerrungen höheren als 3. Grades gemacht worden?

VORTRAGENDER: Nein, weil es bisher nicht gelang, Verstärker mit Verzerrungen hohen Grades ohne Verzerrungen niedrigen Grades zu bauen.

BAERLE, F. C. G. v.: Sind die Grenzen der Hörbarkeit nichtlinearer Verzerrungen gegen Frequenzschwankungen, wie sie bei mechanischer Schallaufzeichnung auftreten, empfindlich?

VORTRAGENDER: Nein.

SPÄLTI, A.: Ist das Gehör für quadratische oder für kubische Verzerrungen empfindlicher?

VORTRAGENDER: Für kubische Verzerrungen bei obertonarmen Klängen, für quadratische Verzerrungen bei obertonreichen Klängen.

ETZOLD, H.: Ist das Gehör für nichtlineare Verzerrungen bei stationären Klängen oder bei nicht-stationären Klängen, wie sie in der Musik vorkommen, empfindlicher?

VORTRAGENDER: Für stationäre Vorgänge, aber der Unterschied ist nicht groß.

STAPPENBECK, H.: Sind Verzerrungen bei ungleichen Amplituden der Primärklänge untersucht? Nach meiner Erfahrung sind die kubischen Verzerrungen am deutlichsten wahrzunehmen, wenn das Amplitudenverhältnis etwa 0,6 : 1 beträgt.

VORTRAGENDER: Solche Untersuchungen haben wir bisher nicht durchgeführt.

THE DESIGN OF HIGH FIDELITY MICROPHONES

by M. L. GAYFORD

Standard Telephones and Cables Ltd. London

Summary. The principles of microphone design are illustrated by the analysis of a modern moving coil microphone. A description is given of the derivation and use of an improved analogous electrical circuit which takes into account the effects of portions of the diaphragm both inside and outside the speech coil diameter.

Sommaire. On montre, en prenant comme exemple l'étude d'un microphone moderne à bobine mobile, les principes du calcul des microphones. On indique comment établir et utiliser un circuit électrique équivalent, du type amélioré, permettant de tenir compte des effets produits par les régions de la membrane qui sont à l'intérieur et à l'extérieur du pourtour de la bobine mobile.

Zusammenfassung. Die Grundprinzipien der Mikrophonkonstruktion werden durch die Analyse eines modernen Tauchspulenmikrophons dargelegt. Herleitung und Anwendung einer verbesserten elektrischen Ersatzschaltung werden beschrieben, bei der die Wirkung der einzelnen Membranteile sowohl innerhalb als auch außerhalb des Tauchspulendurchmessers berücksichtigt werden.

1. Introduction

The analysis of a high-gradè moving coil microphone provides an interesting study of the principles involved in design.

The size and form of the diaphragm system were first decided. Prime requirements are that it must be strong enough to withstand any acoustic shocks likely to occur, whilst the restoring compliance must be linear. The area and mechanical impedance must be such that the available acoustic pressure will drive it at a suitable velocity. Furthermore, the moving system should have only one effective natural period.

The diaphragm is made from a strong aluminium alloy and comprises a rigid domed centre vibrating axially in a compliant surround. The coil is wound from aluminium tape and is cemented directly to the base of the dome. The effective diameter of the diaphragm is about 2.5 cm. The magnet and pole pieces provide a radial flux density of about 10 000 gauss across the coil.

The whole system must be enclosed in a solid case with a protective grille opening before the diaphragm, as the microphone is pressure operated. The mechanical layout of the microphone can be sketched and a functional diagram drawn (Fig. 1). This identifies the parts acting as masses or resistances and those acting as compliances.

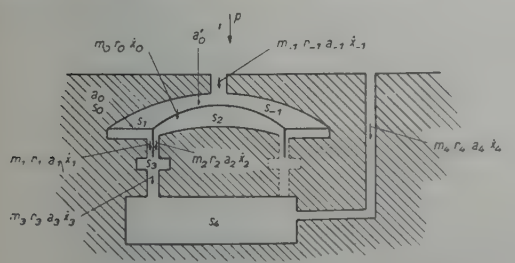


Fig. 1. Functional diagram for the microphone.

2. The analogous electrical circuit

Circuits previously derived for these microphones have been of limited value because the separate acoustic networks coupled to the rear parts of the diaphragm inside and outside the coil diameter were lumped together as one [1]. Some while ago the author derived a circuit, which took account of the two areas of the diaphragm.

The values of the various constants were estimated and these were expressed in analogous acoustic units [2]. The velocities $\dot{x}_0$, $\dot{x}_1$ etc. of the masses were taken as independent co-ordinates and the equations of motion were derived by applying d'Alembert's principle and Lagrange's equation (see Appendix). Use was made of the "direct" analogy, e.g. mass $\equiv$ inductance [3].

The equations also apply to the equivalent electrical circuit (Fig. 2). The resultant acoustic pressure

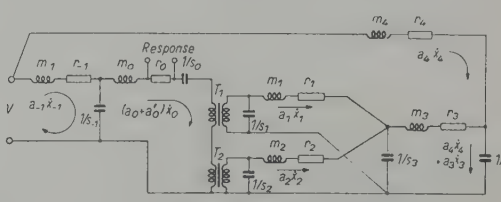


Fig. 2. Analogous electrical circuit.

applied to the microphone openings is represented by the voltage V applied to the terminals of the network, the current through m_0 representing the coil velocity and hence the response of the microphone. The two parts of the diaphragm feeding air flow through the separate slots on each side of the coil indicate two ideal transformers T_1 and T_2 whose transformation ratios are a function of the relative values of the areas of the diaphragm dome and surround.

3. Results given by the analogy

Suitable equivalent electrical values were selected (e.g. 1 millihenry ≡ 1 milligram etc.) and the electrical circuit was set up. Varying the component values, whilst a constant voltage is applied to the input terminals, shows the effect on the response. For instance, it is found that the compliance $1/s_4$ of the case volume and the diaphragm surround compliance $1/s_0$ control the low frequency response, the acoustic resistance r_3 controls the middle frequency response and the diaphragm mass m_0 controls the high frequency response. The front grille m_{-1} and air cavity $1/s_1$ form a damped Helmholtz resonator which aids the extreme top response, and the mass m_4 of the air leak tube interacts with the other elements and aids the response at extreme low frequencies. Judicious experiment enables circuit values to be found which give the best overall response.

A square wave-form voltage is applied to check the transient response of the system.

This lumped-constant analogy ignores any wave propagation effects in cavities of size comparable to the acoustic wavelength. In the actual microphone certain cavities are treated with sound absorbing material to prevent such effects.

The circuit response corresponds to constant acoustic pressure on the microphone opening and, to obtain an estimate of the field response, the diffracting effects of the actual case have to be added as a correction. A porous front baffle, placed before the diaphragm opening, largely compensates for diffraction effects and makes the microphone omnidirectional at all frequencies [1].

4. Conclusions

This analogy has proved valuable in design work and has led to considerable improvement in the response of earlier microphones and facilitated the development of new types. In particular, the importance of giving correct values to the impedance of the cavities etc. under the two parts of the diaphragm has been established.

Acknowledgments are due to Messrs. Standard Telephones and Cables and to many of the authors' colleagues.

Appendix

From Lagrange and d'Alembert

$$\frac{\mathrm{d}}{\mathrm{d}t}\left(\frac{\partial T}{\partial \dot{x}_n}\right) + \frac{\partial F}{\partial \dot{x}_n} + \frac{\partial V}{\partial x_n} = e_n(t),$$

where T, F and V are the kinetic, frictional and potential energy functions respectively. From Fig. 1 expressions for the energy associated with each coordinate can be derived. Differentiation gives the equations of motion. These are seen to be the mesh equations for the equivalent electrical circuit. For the diaphragm mesh:

$$\begin{aligned} &\mathrm{j}\omega m_0 (a_0 + a_0')\dot{x}_0 + r_0 (a_0 + a_0')\dot{x}_0 + \frac{s_0}{\mathrm{j}\omega}(a_0 + a_0')\dot{x}_0 + \\ &+ \frac{s_{-1}}{\mathrm{j}\omega}\left[(a_0 + a_0')\dot{x}_0 - a_{-1}\dot{x}_{-1}\right] + \\ &+ \frac{a_0}{a_0 + a_0'}\frac{s_1}{\mathrm{j}\omega}(a_0\dot{x}_0 - a_1\dot{x}_1) + \\ &+ \frac{a_0'}{a_0 + a_0'}\frac{s_2}{\mathrm{j}\omega}(a_0'\dot{x}_0 - a_2\dot{x}_2) = 0. \end{aligned}$$

Thus two ideal transformers of ratios $\frac{a_0}{a_0 + a_0'}$ and $\frac{a_0'}{a_0 + a_0'}$ are introduced by the dome and the surround.

References;

[1] Marshall, R. N., Romanow, F. F., Bell Syst. tech. J. 15 [1936], 405.
[2] Morse, P. M., Vibration and sound, 2nd Ed. [1948], 233.
[3] Olson, H. F., Dynamical Analogies, 1st Ed. [1943].

MEASUREMENTS OF NON-LINEAR DISTORTION IN LOUDSPEAKERS

by F. Ingerslev

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Summary. A new method for the determination of non-linear distortion in a loudspeaker based on measurement of the sound pressures of the new tones generated by the loudspeaker when driven by a stimulus of two tones of sinusoidal waveform is described.

Sommaire. On décrit une nouvelle méthode de détermination de la distorsion non linéaire dans un haut-parleur; elle repose sur la mesure de la pression sonore des nouveaux sons qu'émet le haut-parleur quand il est excité par deux sons sinusoïdaux.

Zusammenfassung. Es wird eine neue Methode zur Bestimmung nichtlinearer Verzerrungen in Lautsprechern beschrieben, die auf der Messung der Töne beruht, die ein von zwei Sinustönen erregter Lautsprecher neu erzeugt.

Non-linear distortion in loudspeakers has usually been disregarded in the past, the attention being concentrated on linear distortion. It will be shown in this paper that many loudspeakers have non-linear distortion which should not be disregarded, and a new method for measurement of non-linear distortion in loudspeakers will be described.

The non-linear distortion may be due to excessive amplitudes of the diaphragm where Hooke's law no longer applies, or to inhomogeneity of the flux density through which the voice coil moves.

The old method for objective determination of the non-linear distortion in a system is based on measurement of the harmonics produced by the system when driven by a stimulus of sinusoidal waveform. This method is useful when the non-linear distortion changes slowly with frequency, but big changes in harmonic distortion even for small changes in frequency occur in a loudspeaker and a satisfactory picture of the harmonic distortion produced by a loudspeaker can therefore only be obtained by continuous recording of the distortion through the whole frequency range. Commercially manufactured instruments suitable for such recordings are not avaible on the market at present.

There are, however, other reasons for looking for a new method for continuous recording of the non-linear distortion in loudspeakers. The harmonic distortion measurements do not give a satisfactory expression for the non-linear distortion in the upper part of the frequency-band radiated by the loudspeaker, and the requirements for low content of harmonic components in the stimulus are sometimes rather difficult to fulfil in certain frequency ranges.

The new method for measurement of the non-linear distortion in a loudspeaker is based on measurements of the sound pressures of the new tones generated by the loudspeaker when driven by a stimulus of two tones of sinusoidal waveform. One of the tones has a low fixed frequency, often the low resonant frequency, the other tone has a high continuously varying frequency. The choice of a tone having a low, fixed frequency involves large amplitudes of the membrane and consequently considerable distortion. The amplitude of the sound pressure of each of the new tones produced by the loudspeaker driven by a stimulus of two tones is expressed separately in per cent of the amplitude of the tone with the high frequency. The corresponding ratio in dB can also be used.

Fig. 1 shows a block diagram of the apparatus used for measurement of non-linear distortion. It is in principle a wave-analyser and two extra oscillators. The wave-analyser contains an oscillator (no. 1), the frequency of which can be varied from 50 000 to 34 000 c/s. Oscillator 2 has a frequency which can be adjusted to (50 000 $\pm$ 250) c/s. The adjustment of the frequency is made before the start of the measurement and the adjustment is then kept during the measurement. Oscillator no. 3 has a frequency of 50 000 c/s. The following notations are used for

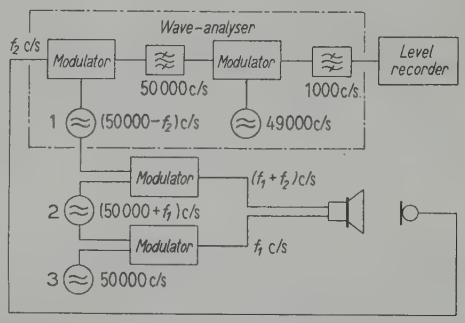


Fig. 1. Block diagram of apparatus used for measurement of non-linear distortion.

the difference frequencies between the frequencies of the various oscillators:

oscillator 3 and 1: f_2,
„ 3 „ 2: f_1 (low fixed frequency),
„ 1 „ 2: $(f_1 + f_2)$ (continuously varying frequency).

The tones having the frequencies f_1 and $(f_1 + f_2)$ are used as the stimulus for the loudspeaker. The wave-analyser is automatically adjusted to the difference frequency or the sum frequency. It will be the difference frequency when the frequency of oscillator 2 is above 50 000 c/s and the sum frequency when the frequency of this oscillator is below 50 000 c/s. It is also possible to arrange automatic adjustment to tones of higher orders if the tone with the low frequency is supplied from an external generator.

The conditions used when measuring the non-linear distortion should be considered carefully. The voltage applied to the moving coil depends partly on the size of the loudspeaker and partly on the purpose for which the loudspeaker is to be used. It has been found better to use the same voltage for both tones, and not a level of the tone with the high frequency which is 12 dB below the level of the tone with the low frequency as is often the case when measuring non-linear distortion in amplifiers.

Fig. 2 shows the result of a measurement. The upper curve shows the ordinary sound pressure response of a loudspeaker and the lower curve shows the sound pressure of the difference tone produced by the loudspeaker when the loudspeaker is driven by a stimulus of two tones, one having a fixed frequency of 90 c/s and the other having a continuously varying frequency. The amplification of the apparatus used when measuring the sound pressure of the difference tone was 30 dB higher than the ampli-

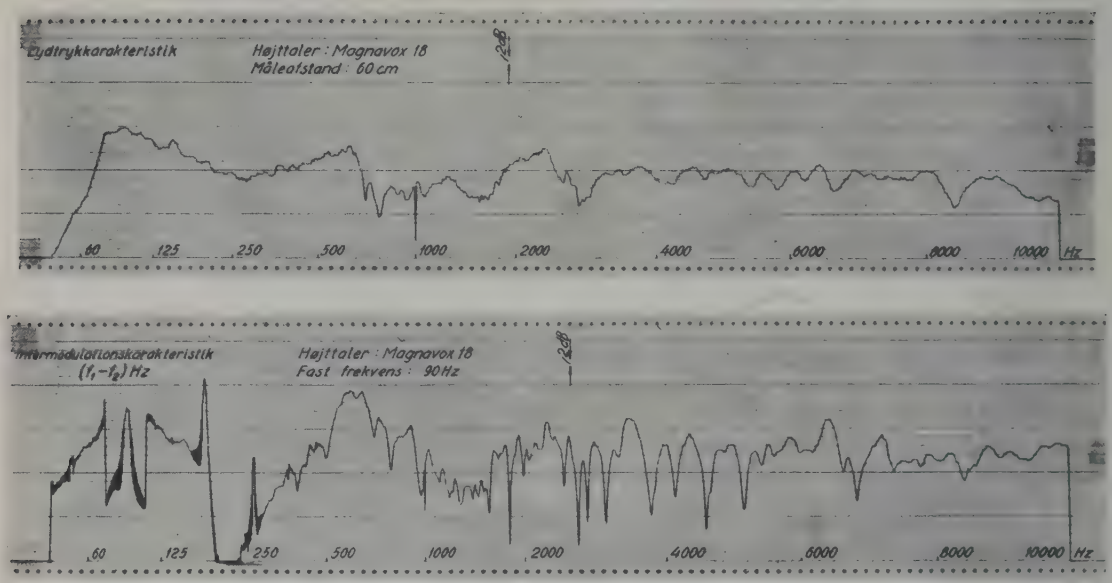


Fig. 2. The upper curve shows the sound pressure response of a loudspeaker. The lower curve shows the sound pressure of the difference tone produced by the loudspeaker when the loudspeaker is driven by a stimulus of two tones, one tone having a fixed frequency of 90 c/s and the other having a continuously varying frequency.

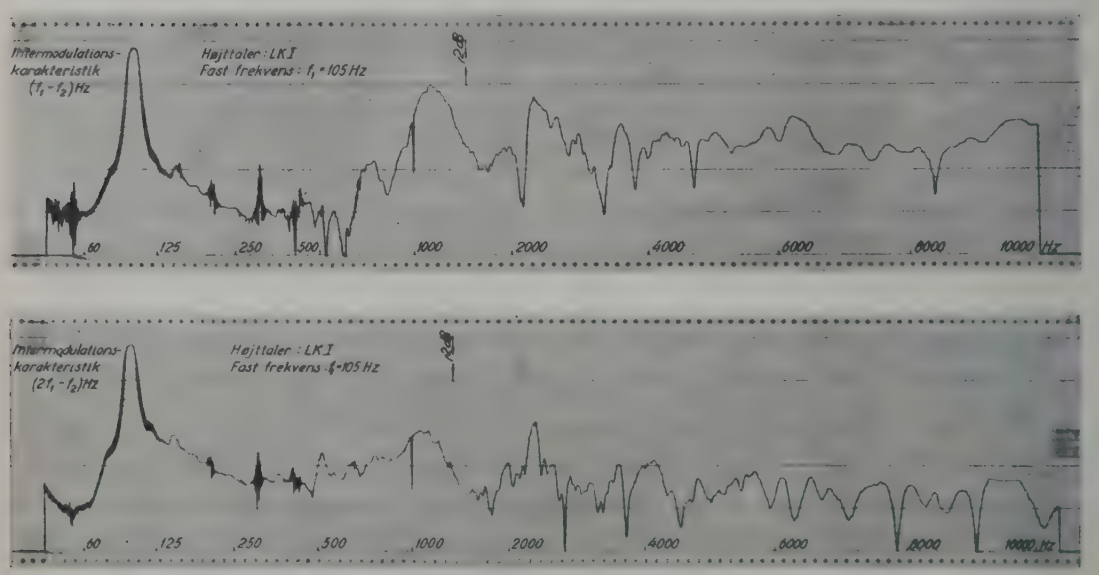


Fig. 3. The sound pressure levels of the difference tone of first order (above) and of second order (below) as a function of frequency.

fication used when measuring the sound pressure response. It is seen that in this case the sound pressure of the difference tone is on the average approximately only 30 dB below the sound pressure of the high frequency.

The upper curve in Fig. 3 shows the sound pressure levels of the difference tone of first order ($f_2 - f_1$) and the lower curve shows the difference tone of second order ($f_2 - 2f_1$). The difference tone of first order has in this case the higher sound pressure level above 1000 c/s.

The lower curve in Fig. 2 shows a pronounced peek in the intensity of the difference tone at 600 ··· 700 c/s. This is the case for many loudspeakers and it usually corresponds with an irregularity in the sound pressure response in the same frequency range. Viscous material applied to the rim of the diaphragm often reduces the magnitude of the non-linear distortion in the frequency range 600 ··· 800 c/s.

The method described above using a stimulus of two tones, one tone having a low fixed frequency, the other having a high continuously varying frequency seems to be a useful method when measuring non-linear distortion in loudspeakers.

Discussion

KNOWLES, H.: We have found it necessary in similar high-frequency glide tone difference frequency measurements to vary the high-frequency power to conform to the peak power demand on speech and music.

LECTURER: The distortion is expressed as the ratio of the sound pressure of the individual difference tones (summation tones) to the sound pressure of the high frequency. Under these conditions a variation of the high-frequency power is of minor influence.

VERMEULEN, R.: Am I right that the difference tone does not give any indication of symmetrical distortion, e.g. cubic distortion?

LECTURER: The difference tones of second order are due to symmetrical distortion.

CHAPMAN, R.: How are second order difference tones measured, and are the curves of difference frequency shown the actual intensity levels of these tones or the difference in intensity levels of these and the high frequency fundamental?

LECTURER: The second order difference tones were measured using an external low-frequency oscillator. The curves show the actual intensity levels of the difference tones.

BRÜEL, P. V.: What is the limiting factor which made it necessary to overload a loudspeaker for using the method. I think it would be nice to develop the method so that it would also be possible to measure the speaker with half or normal load.

LECTURER: The loudspeaker was usually not overloaded. It will, however, be difficult to record with high accuracy the non-linear distortion produced by very good loudspeakers using only half the normal load or even a lower value because the lowest measurable level of the recorded difference and summation tone was 45 dB below the level of the high frequency.

UN NOUVEAU TRADUCTEUR ELECTROACOUSTIQUE: L'IONOPHONE

par S. KLEIN
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Sommaire. Description et applications pour la reproduction musicale et les ultra-sons du nouveau haut-parleur français sans membrane.

Zusammenfassung. Beschreibung des neuen französischen Lautsprechers ohne Membran und seine Anwendungen zur Musikwiedergabe und zur Ultraschallerzeugung.

Summary. A description of a new French loudspeaker without membrane and its uses in music reproduction and for ultrasonics.

1. Introduction

Le classique haut-parleur à membrane est, sans aucun doute, le maillon le plus imparfait de toute la chaine de reproduction sonore. Et ce n'est pas d'hier que de nombreux chercheurs ont tenté de se libérer de l'inertie de la matière, l'idéal étant évidemment d'agir directement si possible sur les molécules d'air.

Ne parlons que pour mémoire du haut-parleur électro-pneumatique, dans lequel une sorte de robinet commandé électriquement délivrait en plus ou moins grande quantité l'air comprimé enfermé dans une bouteille. Un tel appareil restait matériel dans son mécanisme de commande, et par conséquent sujet aux mêmes inerties que le haut-parleur à membrane. Un autre moyen de secouer au rythme voulu les molécules atmosphériques, consistait à tenter de donner à celles-ci une polarisation électrique. C'est ce qu'on sait faire, au moyen de l'ionisation. Déjà quelques expérimentateurs avaient remarqué que si un arc électrique est alimenté en courant modulé, l'arc devient la source d'ondes sonores. Mais le rendement électrique est déplorable, et la tonalité très mauvaise. D'autres essais obtenus en plaçant une flamme entre les armatures d'un con-

densateur ne furent guère plus heureux, le manque de puissance acoustique étant toujours l'obstacle majeur.

En 1946, l'auteur présentait à l'Académie des Sciences et à la Société des Radioélectriciens un haut-parleur ionique. Il s'agissait en somme de la réplique ionique de la triode électronique: un filament chauffant portait à haute température une électrode recouverte de substance fortement émettrice d'ions; un cylindre perforé tenait le rôle de grille et modulait le flux ionique; une troisième électrode, portée à un fort potentiel négatif, jouait le rôle de l'anode dans le tube à vide. Le rendement n'était pas extraordinaire puisqu'il fallait 500 W de puissance d'alimentation pour produire un son faiblement audible dans une pièce de dimensions moyennes; mais la fidélité était bonne et il semblait que l'on était sur la bonne voie.

Pour améliorer ces résultats, il fallait d'abord accroître l'émission ionique. L'auteur y travailla de nombreuses années et parvint, à l'aide conjuguée de l'emploi d'un chauffage H.F. et d'un mélange émissif ainsi composé:

mousse de platine	50%
phosphate d'aluminium	40%
iridium pulvérulent	5%
graphite	5%

à obtenir une puissante source d'ions.

Il fallait encore adapter les impédances acoustiques et ce fut le classique pavillon exponentiel qui fournit la solution. En fin de compte, le haut-parleur ionique version 1951 pouvait être représenté comme la coupe de la fig. 1.

2. Fonctionnement de l'ionophone

L'ionophone est essentiellement constitué par un condensateur coaxial dont l'armature interne est un fil se terminant par une sphère de platine, noyé dans un cylindre de quartz.

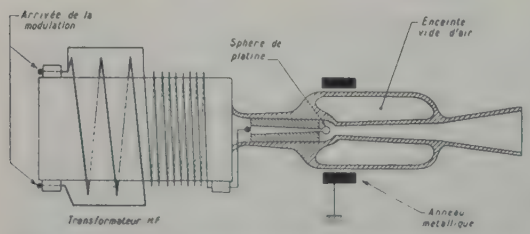


Fig. 1. La cellule thermoionique couplée à son transformateur haute-fréquence.

L'autre électrode est un anneau métallique concentrique. Le cylindre de quartz central est isolé électriquement et thermiquement au moyen d'une enceinte vide d'air. La paroi intérieure de cette enceinte amorce le départ du pavillon exponentiel, ce dernier est prolongé hors de la cellule par une trompe métallique par exemple.

Un signal haute fréquence de très haute tension est appliqué aux bornes de ce condensateur. Cette tension provient du secondaire élévateur d'un transformateur dont le primaire est alimenté, dans l'état actuel de la réalisation, par un câble coaxial provenant d'un générateur de haute fréquence classique. Lorsque la tension H.F. est appliquée, on observe qu'une lueur violette entoure le cylindre de quartz; bientôt, sous l'effet des pertes diélectriques, ce dernier est porté au rouge, et l'appareil est en état de fonctionner. La sphère de platine émet alors un flot d'ions positifs. Si, maintenant, on module en amplitude par un procédé quelconque la très haute tension à haute fréquence, l'ionisation se modifie proportionnellement, il en résulte des variations instantanées du nombre de chocs moléculaires ioniques engendrés dans la cellule, autrement dit de la pression régnant dans le pavillon, d'où naissance d'un son.

3. Limite du fonctionnement

L'ionophone était destiné, primitivement, à la reproduction musicale. C'est donc dans le spectre sonore qu'ont commencé les premières recherches de ses limites. Vers les fréquences élevées, il apparut très rapidement que la cellule ionique battait de loin le haut-parleur à membrane. Quant à la limite inférieure des fréquences reproductibles, elle n'est, en fait, déterminée que par la grandeur que l'on peut donner au pavillon exponentiel. Si ce dernier peut recevoir un allongement tel que son ouverture puisse être de l'ordre de quelques mètres, rien ne s'oppose à ce que des fréquences aussi faibles que 20 ou 30 Hz puissent être retransmises.

Par contre, si, pour des raisons d'encombrement (emploi dans un récepteur de radio domestique par exemple) on est contraint d'adopter un pavillon de dimensions réduites, il est certain que la reproduction des fréquences basses sera défaillante. L'on résout actuellement la difficulté en associant au haut-parleur ionique un haut-parleur classique de grand diamètre (boomer), lequel est chargé par un filtre de séparation classique de la reproduction des notes basses. Cette solution est d'ailleurs tout à fait satisfaisante, étant donné que les progrès de la technique actuelle des haut-parleurs sont tels que l'on sait faire fonctionner assez correctement un haut-parleur de grand diamètre pour des fréquences ne dépassant pas 300 Hz.

4. Fonctionnement aux fréquences très élevées

Les courbes de réponse obtenues dans le domaine des fréquences musicales montrent qu'au-delà de la fréquence de coupure du pavillon, aucune atténua-

tion ne se produit jusqu'à des fréquences très élevées.

Bien que sans intérêt pour la reproduction musicale seule, il était extrêmement intéressant d'observer le comportement à des fréquences élevées; on s'aperçut alors que le fonctionnement restait possible, non seulement pour les fréquences ultra-sonores proches, mais aussi pour des fréquences pouvant atteindre le mégacycle par seconde. C'était là une découverte extrêmement importante, étant donné le nombre sans cesse grandissant des applications des ultra-sons et étant donné d'autre part les difficultés que l'on a pour produire, avec une puissance importante, des vibrations acoustiques de fréquence très grande.

5. Les ultra-sons

Tous les générateurs classiques d'ultra-sons (les cristaux piezoélectriques, les barreaux magnétostrictifs) ont en commun deux inconvénients: d'une part, ce sont des générateurs accordés, conçus pour travailler sur une seule fréquence ou ses harmoniques; d'autre part la transmission ne s'effectue correctement que dans des milieux liquides ou solides, alors qu'il peut être intéressant, dans bien des cas, d'envisager l'ébranlement des molécules gazeuses. Pour cette dernière application la cellule ionique est idéale. On a pu ainsi vérifier, que, dans l'air, des intensités aussi grandes que 150 dB au dessus de $2 \cdot 10^{-5}$ N/m^2 pouvaient être obtenues sans difficulté (rappelons pour comparaison qu'un moteur d'avion à réaction, à quelques mètres, correspond à un bruit d'environ 140 dB). La portée dans l'air dépend évidemment de la fréquence, car on sait que les ultrasons sont rapidement amortis dans un milieu gazeux et cela d'autant plus que la fréquence est grande. Toutefois, des portées d'une centaine de mètres sont faciles à réaliser pour une fréquence de 30 kHz par exemple. Des ultra-sons de 3 MHz ont pu traverser plusieurs centimètres d'air.

Cependant, pour concurrencer victorieusement les autres générateurs d'ultra-sons, l'ionophone devait être capable de transmettre les vibrations à des milieux liquides ou solides. Cela posait une délicate question d'adaptation des impédances. Ce problème a été résolu très simplement par l'emploi d'un matériau intermédiaire, d'élasticité et de densité telles que l'adaptation correcte pouvait avoir lieu avec un rendement élevé. Pratiquement, on emploie par exemple un tampon de mousse de caoutchouc placé contre l'ouverture du pavillon exponentiel de la cellule.

6. Applications aux ultra-sons

On a dit plus haut qu'une caractéristique principale de la cellule ionique était son apériodicité. Dans presque tous les domaines où les ultrasons sont mis en oeuvre, cela ne manquera pas de présenter d'immenses avantages.

En médecine, par exemple, on se contente, jusqu'à présent, d'expérimenter sur un certain nombre de fréquences fixes. Il est certain que des découvertes intéressantes peuvent résulter de l'exploration méthodique du spectre ultra-sonore et cela sans discontinuité. Des émissions d'ultra-sons modulées en fréquence sont également concevables, ce qui ne pouvait se faire au moyen des générateurs magnétostrictifs ou à quartz que d'une facon extrêmement étroite. Il est facile d'imaginer les commodités qu'offrent ces perspectives pour les liaisons aériennes à courte distance, où les liaisons sous-marines, le repérage à distance, la télécommande, etc.

En utilisant l'ionophone comme émetteur d'ultrason dans l'air, une première réalisation pratique a été inaugurée récemment à Paris, consistant en la transmission de paroles sur porteuse ultra-sonore. Il s'agissait dans ce cas concret, de transmettre dans la salle des conférences de l'UNESCO des traductions dans diverses langues des discours prononcés. Les traductions s'effectuent de la facon habituelle; après quoi les courants de basse fréquence modulent des générateurs, suivis chacun d'une cellule ionique, et disposés au plafond de la salle de conférence. Les réceptions s'effectuent au moyen d'un casque téléphonique. Chaque auditeur porte un tel casque sur la bride duquel est fixé le récepteur proprement dit, qui comprend un microphone à cristal, suivi d'un amplificateur alimenté par batterie. Les fréquences porteuses sont de l'ordre de 36 kHz. Un seul tube amplificateur du type subminiature est nécessaire pour détecter et transmettre aux écouteurs la puissance basse fréquence. L'accord sur une des langues est effectué au moyen de trois boutons poussoirs branchant des capacités différentes aux bornes d'un circuit oscillant bobiné sur noyau magnétique. La commodité d'usage est extrême; de plus, le secret des transmissions est absolument respecté puisque aucun signal ne parvient à l'extérieur de la salle.

ARTIFICIAL ACOUSTICAL ENVIRONMENT CONTROL

by H. S. KNOWLES

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Summary. The acoustic field at the listener differs markedly in a concert hall and in an outdoor amphitheatre. To provide a sound field outdoors approximating to that of a hall an array of loudspeakers was suspended over and around an outdoor area and supplied with multiply delayed, attenuated and spectrally modified signals. Listeners showed a significant preference for the simulated indoor acoustical environment even when presented with a visible outdoor environment.

Sommaire. Le champ sonore à l'endroit où se trouve un auditeur n'est pas le même suivant que l'auditeur est dans une salle de concert ou dans un amphithéâtre en plein air. Pour produire en plein air un champ sonore qui reproduise sensiblement celui existant en salle, on a suspendu une série de haut-parleurs au-dessus et autour d'un terrain en plein air et on les a alimentés avec des signaux présentant plus ou moins de retard, d'affaiblissement, et de modification de leur spectre. Les auditeurs ont montré une préférence marquée par cette simulation d'une ambiance acoustique intérieure, même quand elle est présentée dans un cadre du plein air.

Zusammenfassung. Das Schallfeld am Ort des Hörers ist in einem Konzertsaal merklich verschieden von dem in einem Freilichttheater. Um das Schallfeld im Freien dem eines Saales anzunähern, wurde eine Lautsprecheranordnung über dem und um das Gebiet im Freien angebracht, die mit verzögerten, abgeschwächten und im Spektrum veränderten Signalen gespeist wurde. Die Hörer bevorzugten auch im Freien die vorgetäuschte akustische „Konzertsaal"-Umgebung.

Historically, in the design of structures for outdoor musical concerts, primary attention has been given to obtaining adequate intensity over the listening area. Since the outdoor area was frequently large, and the ambient and audience noise significant, reflecting surfaces were placed near the performers to make the intensity as high and uniform as possible over the listening area. The reflecting surface also served to reflect sound to the performers thereby providing more nearly the acoustical environment needed to aid and stimulate the performer.

Sound reinforcing systems (SRS) have aided outdoor music reproduction by raising the intensity levels and providing more uniform spatial distribution. The objective has been to have a spatially uniform, linear, frequency and transient distortion-free transfer characteristic from the microphone to the listener. The SRS raises the intensity level but does not significantly alter the characteristics of the sound field.

This sound field differs radically from the indoor sound field found in listener-preferred concert halls where most critical music listening is done. This raises the question whether outdoor listening would be enhanced by providing a simulated indoor acoustical environment. The objective of simulating a preferred indoor acoustical environment raises all the questions, many of them unanswered, that are raised by the entire subject of room acoustics. We can here briefly report only on some of the tests conducted to answer this question and on the major criteria that were considered important for the simulated environment.

All preferred concert halls have reverberation times of the order of 1.3 to 2.0 seconds with a modal value of about 1.5 seconds. The shape of the attack and decay curves lies within moderate limits. Investigators agree that the intensity, direction and time delay of the first few reflections (the characteristics of the first few "images") are important. Preferably without, but certainly with the aid of the visual stimulus of the performer, the listener should identify the sound with the performer and not be able to resolve other acoustic images. The quality should be reasonably uniform throughout the area. The environment should be free of detectable echo, buzz, and flutter.

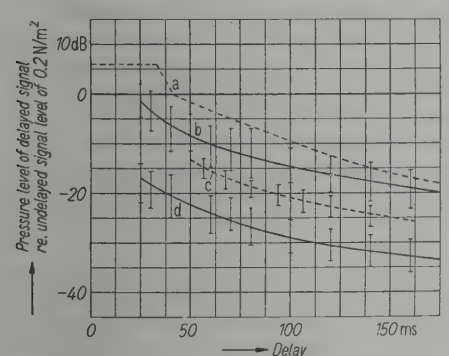
Simulation of reverberation suggests the use of lumped or distributed parameter electromechanical reverberation networks to provide a reverberant signal without modifying the conventional speaker system. These were tried and many listeners preferred the results. The listeners did not however consider that they were immersed in an indoor field. This was due, in part at least, to the difficulty of producing the desired pressure-time pattern at two fixed ear coordinates and to the fact that even if it is produced the listener can detect the space gradient by slight motions of the head and gain clues to the direction of field components.

The simplest method of roughly simulating the important indoor images and the attack and decay appears to be to surround the outdoor listener with an array of speakers supplied by electrical signals so modified that each speaker radiates a wave which corresponds to the wave the simulated room bound-

ary would reflect from that region into the listening area. The proper delay, attenuation (and its frequency dependence) can be approximated by a study of the successive images in (the geometry and boundary nature of) a preferred environment.

For economic reasons it is necessary to limit the number of speakers and one important problem is to determine subjectively the configuration of a minimum array. One way of reducing the number of speakers and roughly simulating the missing ones, is to supply some speakers with a signal delayed or advanced by an amount which simulates the nearest missing or "phantom" speaker in its most critical listening area. The time change so introduced is normally of the order of 20 to 40 milliseconds.

To test the subjective interference of short delays, four free field speech and four dead studio (0.6 s RT) music recordings were made. These magnetic recorded signals were supplied to a delay mechanism which provided 0 ··· 150 ms delay. The initial interference test was limited to six trained listeners (acoustical engineers and musically trained radio programme directors). The maximum, minimum and mean values for these listeners and four speech samples (syllable rates of 4.5 to 5.5/s) are shown in the figure. A single delayed signal mixed electrically with the undelayed and fed to a single isolated free field source was used. The intensity levels were those found desirable in earlier outdoor work and lie nearer cinema than indoor concert levels. The curve designations b ··· d indicate (see legend to figure)



The pressure level difference of a delayed signal and the undelayed signal of 0.2 N/m² according to:
a) Bolt and Doak's 10% disturbance criterion, based on Haas' experiments;
b) direct subjective estimation of "definitely objectionable;
c) direct subjective estimation of "echo identified";
d) direct subjective estimation of "minimum perceptible".

the questions asked the listeners (curve a for comparison). The level of the interfering signal required to produce the specified subjective estimate rises as the time delay of the interfering signal is reduced from 170 to 25 ms.

On music the mean interference pressure levels were substantially higher. The deviations were much greater and depended markedly on the programme material (tempo, staccato, etc.) and on the listener, so that larger listener and music samples are needed to give statistically significant results. The speech values are given because although the samples are small the deviations are moderate and they give high interference values which provide safe criteria for music. For comparison Bolt and Doak's [1] "10% disturbance" curve based on Haas' [2] data is shown. This corresponds roughly to the "definitely objectionable delay" region of our figure. The minimum perceptible intensity values were found to rise in the presence of other interfering signals of higher intensity much as one would predict. The presence of other signals and the lower correlation normally obtained in acoustic interference may account for the higher interference-intensity tolerance reported by Haas, Bolt and Doak.

An experimental array was tested in which the loudspeakers were approximately 8 m above ground, 12.5 m apart across the area and 16 m apart along the area. A rectangular array was used for structural reasons. The electrical delay to each speaker did not in general correspond to the acoustic transit time from the performer to the loudspeaker. A 20 speaker array was effectively increased in size by a factor of four by using two large perpendicular high impedance walls to provide reflections. A theatre type loudspeaker system was mounted at the intersection of the two walls and ground. The recordings used to simulate music played in an outdoor shell were recorded in a 0.6 s RT sound stage. Sixteen adjustable delay circuits, some with feed-back, provided complex delay patterns.

With the complete system adjusted to simulate a preferred 1.5 s RT hall each listener was asked to choose between music amplified over a high quality theatre loudspeaker system and music amplified over the simulated room system adjusted to the same loudness and subjective frequency response balance. Of the 960 judgments made by 24 listeners 918 favoured the simulated room reproduction and 42 the straight amplified signal. Choices favouring the straight amplification were nearly all based on a staccato passage which permitted detection of the defects of the delay system and finite speaker separation. The defects are primarily the lack of the damped sinusoidal terms corresponding to the transient room mode responses and the disordered part of the attack and decay. With more circuit complication these can be added using reverberation circuit techniques.

Such an arrangement permits the size, shape and boundary impedance of the simulated room to be varied by a substantial factor by purely electrical means. This permits the "room" to be fitted to the music or source material. The ratio of direct incident or only slightly delayed sound to more delayed may differ for different sound sources at the same time. Improved "intimacy" was obtained by reducing the RT on a solo voice and leaving the normal RT on music. The artificial room also provides a tool for the study of room acoustics.

While the initial tests were too limited to permit generalization they were conclusive enough to indicate that most listeners prefer an indoor acoustical environment when listening to music outdoors. A supplementary test was later completed in the spring of 1950 of a normal size installation in which however the listeners near the orchestra were covered by a ceiling (no walls) and in which the remote outdoor listeners were supplied with delayed sound. The combined test results were promising enough so that a full scale installation of the completely simulated environment is planned.

References

[1] Bolt, R. H. and Doak, P. E., J. acoust. Soc. Amer. 22 [1950], 4.
[2] Haas, H., Acustica 1 [1951], 2.

KONDENSATOR-HOCHTONLAUTSPRECHER UND KONDENSATOR-MIKROPHONE MIT FESTEM DIELEKTRIKUM

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Zusammenfassung. Mit gerillten oder gelochten Gegenelektroden und auf ihnen aufliegenden, gespannten und außen metallisierten Kunststoff-Folien kann man Kondensator-Lautsprecher bauen, deren Frequenzkurve in erster Linie durch die Form der Gegenelektrode gegeben ist, und deren Wirkungsgrad trotz geringer Gleichspannung den von dynamischen Lautsprechern oberhalb von 5 kHz überschreitet. Kondensator-Mikrophone dieser Art können fast dieselbe zeitlich konstante Empfindlichkeit im Hörbereich wie die mit Luftdielektrikum haben, aber eine wesentlich größere im Ultraschallbereich.

Summary. Counter-electrodes with grooves or holes and having stretched over them thin layers of plastic foil metallised on the outside can serve as condenser loudspeakers, whose frequency response depends on the form of the counter-electrode and whose efficiency even on low potential exceeds that of dynamic loudspeakers above 5 kc/s. Condenser microphones with this speciality have nearly the same sensitivity as those with an air dielectric in the audio range but much greater sensitivity in the ultrasonic range.

Sommaire. En employant des contre-électrodes à trous ou à rainures, sur lesquelles on applique et on tend des feuilles de produit synthétique métallisées à l'extérieur, on peut réaliser des haut-parleurs condensateurs dont la courbe de réponse en fréquence est déterminée surtout par la forme des contre-électrodes, et dont le rendement dépasse, au-delà de 5 kHz, celui des haut-parleurs dynamiques, bien que la tension continue soit faible. Des microphones-condensateurs du même type peuvent avoir, dans la gamme audible, presque la même sensibilité constante dans le temps que les microphones dont le diélectrique est de l'air, mais leur gamme ultrasonore est notablement plus étendue.

Über die mehrjährige Entwicklung von Kondensator-Mikrophonen für Ultraschall und Kondensator-Lautsprechern im III. Physikalischen Institut der Universität Göttingen, die mit Unterstützung von DSIR in London durchgeführt wurde, erscheint demnächst in der Acustica eine ausführliche Veröffentlichung von Kuhl, Schröder und Schodder. Deshalb soll auf eine vollständige Wiedergabe des Referates verzichtet werden, und es seien im folgenden nur die wesentlichen Ergebnisse der Untersuchungen genannt.

Kondensator-Lautsprecher und -Mikrophone, bestehend aus einer nichtleitenden, außen metallisierten Folie, die an vielen Punkten auf den Unebenheiten einer metallischen Gegenelektrode aufliegt, wurden schon 1929 patentiert und später von Sell, der auch einige Mikrophon-Frequenzkurven veröffentlichte, in verschiedenen Ausführungsformen beschrieben. Darauf aufbauend wurden jetzt Mikrophone mit einer oberen Grenzfrequenz von 50 bis 100 kHz entwickelt mit konzentrischen Rillen in der Gegenelektrode und mit außen durch Bedampfung metallisierten Kunststoff-Folien von etwa 10 μm Dicke als Membran. Wenn die Elektroden zwischen den Rillen aufgerauht sind, ist die Empfindlichkeit, aber auch die Gefahr von elektrischen Durchschlägen größer. Die Mikrophone bestehen aus einer Kugel von 35 mm Durchmesser, die ein ebenes System von 13 mm Durchmesser, mit ihrer Oberfläche fast abschließend, enthält. Die untersuchten

ebenen Lautsprecher haben einen Durchmesser von 84 mm.

Die Absoluteichung der Mikrophone geschah bei Frequenzen von 3 bis 200 kHz im freien Schallfeld nach der Reziprozitätsmethode, und zwar mit gleichartigen Systemen als Sender, umkehrbarer Wandler und Mikrophon. Die Summe aller möglichen Meßfehler betrug bis 100 kHz $\pm$ 0,8 dB, der tatsächliche Fehler aber weniger.

Bei Lautsprechern mit (nach hinten geschlossenen) Rillen in der Gegenelektrode steigt der Schalldruck auf der Achse bei konstanter Wechselspannung proportional dem Quadrat der Frequenz an bis zu einer Resonanzfrequenz, die durch die Abmessungen der Rillen, die mechanischen Eigenschaften und die mechanische Spannung der Folie gegeben ist und beliebig hoch in den Ultraschallbereich gelegt werden kann. Als Hochtonlautsprecher für elektroakustische Zwecke sind Lautsprecher mit (hinten offenen) Durchbrechungen einer dünnen Elektrode besser geeignet. Ihr Wirkungsgrad übersteigt bei gleicher Scheinleistung den von üblichen dynamischen Lautsprechern oberhalb von 5 kHz. Der Schalldruck steigt in einem größeren Frequenzbereich etwa proportional der Frequenz f, dann proportional f^2 an und ist schließlich bis zu etwa 50 kHz fast konstant. Da die Frequenzkurve elektrisch leicht entzerrt werden kann und frei von Schwankungen ist, und der Preis dieser Lautsprecher sehr gering ist, haben deutsche Radiofabriken, wahrscheinlich veranlaßt durch diese Untersuchungen, in den letzten Jahren Kondensator-Hochtonlautsprecher für hochwertige Geräte gebaut.

Es lassen sich Mikrophone mit geschlossenen Rillen bauen, deren Empfindlichkeit gleich der von üblichen hochwertigen Kondensatormikrophonen mit Luftdielektrikum ist, d.h. 10^{-2} V/Nm^{-2} (1 mV/μbar), deren obere Grenzfrequenz (3 dB Abfall vom Grenzwert bei tiefen Frequenzen) aber 50 kHz beträgt. Bei einer um 10 dB geringeren Empfindlichkeit kann die Grenzfrequenz unter anderem durch eine höhere mechanische Spannung der Membran auf etwa 200 kHz verschoben werden.

Der von den Lautsprechern erzeugte Schalldruck bzw. die Empfindlichkeit der Mikrophone steigt bei geringer Gleichspannung proportional zu dieser an, erreicht für Frequenzen unterhalb der Resonanzfrequenz ein Maximum bei 150 bis 200 Volt, und fällt bei größeren Spannungen wieder ab. Bei manchen Folien nimmt der Wirkungsgrad bei konstanter Spannung im Laufe der Zeit infolge von Aufladungserscheinungen ab. Dieser Effekt konnte noch nicht ganz geklärt werden, aber es wurden Mittel zur Abhilfe gefunden und auch Folien, bei denen er ohnehin nur wenig auftritt.

Der Klirrfaktor der Lautsprecher entspricht genau dem Wert, den man aus dem Verhältnis von Gleich- und Wechselspannung und aus der Frequenzkurve berechnen kann. Lautsprecher mit (hinten offenen) Durchbrechungen in der Gegenelektrode, bei denen der Anstieg proportional der Frequenz und nicht proportional deren Quadrat ist, sind in dieser Beziehung günstiger. Bei der Reziprozitätseichung ist die Umkehrbarkeit der Wandler wichtig. Sie setzt voraus, daß der erzeugte Schalldruck proportional der angelegten Wechselspannung ist. Diese Bedingung ist für manche Folienmaterialen erfüllt, für andere nicht.

Porenfreie Membranen der angegebenen Dicke sind durchschlagsfest bis zu Spannungen, die ein Vielfaches der Spannung sind, bei der der Wirkungsgrad sein Maximum hat. Bei gerecktem Polystyrol (Styroflex), das sich bisher als günstigstes Material erwiesen hat, ist die Empfindlichkeit gegen große Temperatur- und Feuchtigkeitsänderungen sehr gering. Mikrophone, die mit einer solchen Folie bespannt sind, die durch Schrauben und zusätzliches Kleben befestigt ist, und deren Gegenelektrode in ihrer Lage fixiert ist, zeigten bei mehreren Messungen über vier Monate eine innerhalb der Meßgenauigkeit der Eichung von $\pm$ 0,8 dB konstante Empfindlichkeit im ganzen gemessenen Frequenzbereich.

Diskussion

HOLLE, W.: Ist bei den Messungen der Stabilität von Mikrophonen mit Polystyrol-Membranen die Polarisationsspannung während der vier Monate ständig angelegt gewesen?

VORTRAGENDER: Die Spannung wurde nur in Abständen von vier Wochen bei der Eichung der Mikrophone angelegt.

BUCHMANN, G.: Sie sagten, daß sich der Klirrfaktor aus den elektrischen Werten genau berechnen lasse. Die mechanischen Eigenschaften der Membran müssen aber doch auch eine Rolle spielen. Nach meinen Erfahrungen kann der Klirrfaktor bei Lautsprechern dieser Art recht erheblich sein.

VORTRAGENDER: Die Übereinstimmung mit der Rechnung gilt für kleine Auslenkungen der Membran, wie sie entsprechend der Amplitudenstatistik bei normaler Modulation bei sehr hohen Frequenzen meistens nur vorkommen.

LA CONQUE ÉLIPSON

UN TRANSFORMATEUR DE RAYONNEMENT ACOUSTIQUE

par R. LEHMANN

Centre National d'Études des Télécommunications, Paris

Sommaire. Après avoir rappelé les difficultés d'utilisation, dans des conditions optima, d'un haut-parleur ordinaire, l'auteur décrit un nouveau dispositif, appelé "conque" par le constructeur, permettant d'améliorer les conditions de rayonnement des transducteurs généralement utilisés. La conque se compose de deux parties distinctes: a) le réflecteur, b) le résonateur. Chacune de ces parties est décrite du point de vue construction et du point de vue propriétés acoustiques. Pour terminer, sont donnés de nombreux résultats objectifs et subjectifs.

Zusammenfassung. Nach einer Darlegung der Schwierigkeiten bei der Verwendung von Lautsprechern unter optimalen Bedingungen beschreibt der Autor eine neue Anordnung, vom Konstrukteur „Muschel" („conque") genannt, die es gestattet, die Abstrahlungsbedingungen der im allgemeinen benutzten Lautsprecher zu verbessern. Die „Muschel" setzt sich aus zwei verschiedenen Teilen zusammen: dem Reflektor und dem Resonanz-Hohlraum. Jeder der Teile wird vom Gesichtspunkt der Konstruktion und der akustischen Eigenschaften behandelt. Zum Schluß werden zahlreiche objektive und subjektive Resultate angegeben.

Summary. After emphasizing the difficulties of using an ordinary loudspeaker under optimum conditions, the author describes a new type, called "shell", which gives an improvement over the commonly used transducers. The shell consists of two distinct parts, reflector and resonant enclosure each of which is described as to its construction and acoustic properties. Data are given.

1. Généralités

L'utilisation maximum d'un haut-parleur à membrane et son montage à l'intérieur d'une enceinte présentent des difficultés. Le dispositif, appelé conque "Élipson", que nous allons exposer met en oeuvre des moyens inhabituels et les résultats obtenus donnent subjectivement une impression de relief sonore.

Ce système vise à obtenir des résultats de bonne qualité à partir d'un seul haut-parleur et réalise, d'une certaine manière, une extension de la répartition spatiale de l'énergie. La conque Élipson joue le rôle d'un transformateur de rayonnement. Elle a pour effet de concentrer, dans un cône sonore utile, l'énergie avant rayonnée par un haut-parleur à membrane.

2. Description de la conque "Élipson"

Ce dispositif est réalisé en deux parties distinctes:

α) Le réflecteur, constitué par une portion d'ellipsoïde allongé qui assure la concentration et la diffusion du son pour toutes les fréquences supérieures à 800 Hz.

β) Un résonateur basé sur les propriétés du résonateur de HELMHOLTZ.

a) *Le réflecteur*

On sait que les ondes sonores subissent, comme les ondes lumineuses, des phénomènes de réflexion, réfraction, et diffraction. L'idée directrice de M. LÉON, réalisateur de la conque, lors de ses travaux, fut d'appliquer à l'acoustique certaines caractéristiques des miroirs courbes, bien connues en optique. Cependant, certaines limitations interviennent par suite de la longueur relativement grande des ondes sonores. En première approximation, on peut admettre que les lois classiques de la réflexion sont valables tant que la longueur d'onde du son émis demeure inférieure à la plus grande des dimensions des miroirs.

Il existe donc, du fait même des dites dimensions, une fréquence de coupure au-dessous de laquelle le réflecteur perd son efficacité.

Dans le dispositif que nous présentons, le sommet du cône du haut-parleur est placé au foyer F de l'ellipsoïde. L'axe géométrique du haut-parleur passe par le point B, sommet du petit axe de l'ellipse méridienne située dans le plan symétrique de l'ensemble (Fig. 1).

Le cercle de base de la membrane s'appuie également sur la même ellipse méridienne. Cette règle pratique n'a, d'ailleurs, pas un caractère impératif.

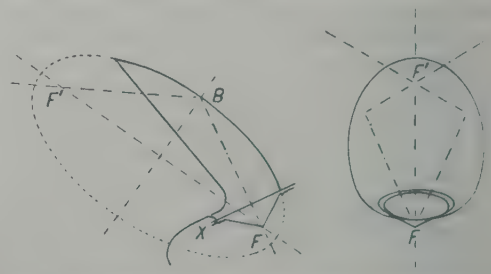


Fig. 1. Disposition du haut-parleur dans le réflecteur de la conque Élipson.
Gauche: Section par le plan symétrique.
Droite: Vue de gauche

En pratique, la portion d'ellipsoïde est limitée à l'avant par une section plane perpendiculaire au plan de symétrie et sensiblement parallèle à la direction

de FB, se raccordant à la partie inférieure avec les parois extérieures de l'enceinte résonante.

L'explication du fonctionnement de la conque ne peut être que schématique. Une étude géométrique, que nous ne présenterons pas mais qui a été faite, démontre le comportement du dispositif.

D'après les propriétés de l'ellipse, tout rayon sonore issu du foyer F se trouvant dans un plan méridien de la surface, y demeure. De plus, ce rayon passe, après réflexion, par le second foyer F'. Le foyer de concentration F' se trouve en avant de la conque, et pour l'auditeur placé dans le cône utile rayonné, l'ensemble des sons réfléchis de fréquences supérieures à 800 Hz, semble émis depuis ce foyer.

Deux remarques sont à faire en ce qui concerne la nature du faisceau sonore émis après réflexion:

α) Le faisceau est un cône dont l'angle d'ouverture est fonction des caractéristiques du miroir elliptique. Dans la pratique, pour une conque de sonorisation, l'ouverture choisie est 45°; pour une écoute d'intérieur, elle est de 60°. Les formes obtenues sont respectivement celles des figures 2 et 3.

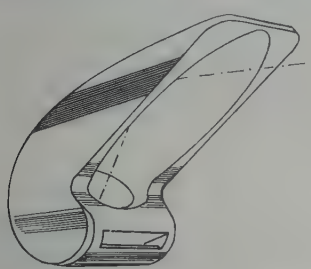


Fig. 2. Forme d'une conque ayant une ouverture de 60°.

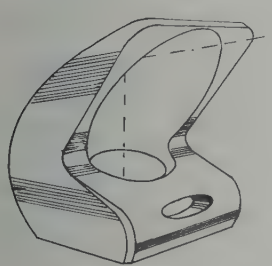


Fig. 3. Forme d'une conque ayant une ouverture de 45°.

β) Le faisceau n'est pas homogène en intensité. Le diagramme de répartition dans un plan vertical est sensiblement celui de la figure 4. La figure 5

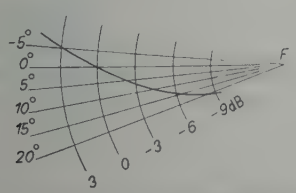


Fig. 4. Caractéristique de directivité de la conque.

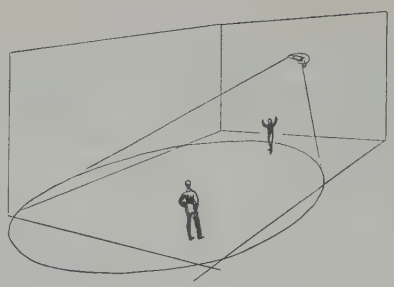


Fig. 5. Obtention de l'équipuissance sonore dans un local.

met en évidence l'équipuissance sonore obtenue dans un local déterminé.

b) *Le résonateur*

Nous venons de voir comment la conque Élipson réalise une meilleure répartition spatiale des sons aigus. Il reste à examiner par quels procédés sont reproduites les fréquences basses. À cette fin, l'arrière de la membrane du haut-parleur est chargée acoustiquement par un résonateur muni d'un évent qui se prolonge à l'intérieur de la cavité (Fig. 6).

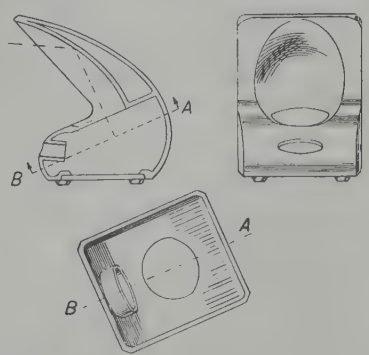


Fig. 6. Constitution de l'enceinte résonante d'une conque Élipson.

Quoique très discuté, ce type de baffle permet un excellent rendement dans le grave sans exiger un volume prohibitif. Une particularité de ce montage réside dans le fait que l'enceinte résonante est constituée par deux cavités communiquant entre elles par une cloison dans laquelle ont été pratiquées des fentes; leur largeur est petite par rapport à l'épaisseur de la cloison. Ce système permet un amortissement variable du résonateur et assure une égalisation du rendement acoustique dans les fréquences basses (Fig. 7).

L'utilisation simultanée du réflecteur et du résonateur constitue un système émissif complexe, les fréquences élevées étant émises au foyer extérieur F' du réflecteur et les fréquences les plus basses au débouché même de la membrane, en X (Fig. 1). La totalité du registre sonore s'étale dans l'espace entre

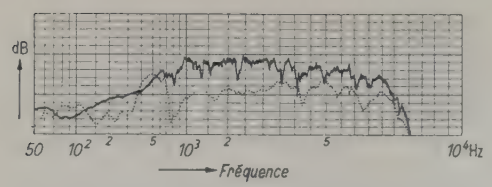


Fig. 7. Égalisation du rendement dans les fréquences basses par le résonateur; une division verticale égale 2 dB.
----- haut-parleur à membrane de 21 cm avec baffle de 1 m,
—— le même haut-parleur avec la conque Élipson.

X et F'. L'expérience d'un amorçage LARSEN en est la preuve.

3. Résultats objectifs

Les conditions théoriques sont vérifiées avec une bonne approximation comme le montrent les résultats suivants:

a) *Courbe de réponse.* Dans la figure 7, on peut remarquer la régularité d'allure de cette courbe, pour l'ensemble des fréquences aiguës. Avec le résonateur, la courbe de réponse est régularisée jusqu'à la fréquence de résonance du haut-parleur. Comme il est facile de le prévoir, les résultats sont fonction de la qualité propre du haut-parleur utilisé.

b) *Courbe de directivité.* Une courbe de directivité dans un plan de symétrie (plan vertical) du diffuseur est donnée dans la figure 8a pour la fréquence 3000 Hz; le microphone de mesure décrit un cercle centré sur F'. On note une dissymétrie par rapport à l'axe du cône, prévisible d'après les propriétés géométriques du rayonnement.

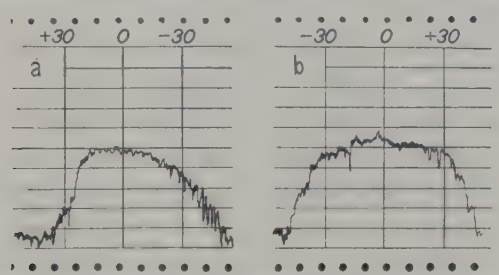


Fig. 8. Caractéristiques de directivité de la conque Élipson; une division verticale égale 2 dB.
a) dans le plan vertical,
b) dans le plan "horizontal".

Une courbe de directivité dans un plan contenant l'axe du cône de rayonnement utile et orthogonal au plan de symétrie (plan "horizontal") du diffuseur, est donnée dans la figure 8b, à 3000 Hz; le microphone de mesure décrit encore un cercle centré sur F'. On notera la constance satisfaisante de l'intensité du rayonnement à l'intérieur du cône utile et son affaiblissement abrupt en dehors de celui-ci.

c) *Rendement.* En moyenne, il a été constaté qu'un haut-parleur monté dans une conque Élipson avait un rendement supérieur de 3 à 4 dB à celui obtenu lorsque le haut-parleur est monté sur un écran plan.

4. Résultats subjectifs

a) *Relief sonore et impression de présence.* Si on alimente le diffuseur Élipson par un oscillateur, l'auditeur constate un déplacement apparent de la source au fur et à mesure que l'on parcourt le spectre du grave à l'aigu. On obtient un véritable étalement en profondeur de la surface émettrice qui s'oppose à la quasi-ponctualité du haut-parleur classique. Toutefois, par la concentration de l'énergie sonore en une plage bien définie à l'intérieur de laquelle la répartition est homogène, on donne à l'auditeur une certaine possibilité de repérage directionnel, d'où un accroissement d'intelligibilité de la parole et la création d'un effet "de présence".

b) *Reproduction des transitoires.* À l'audition, la reproduction des transitoires apparait meilleure avec la conque qu'avec un haut-parleur monté sur baffle plan.

5. Conclusions

Tenant compte des limitations imposées dès le départ, les résultats obtenus apparaissent comme très intéressants.

Dans le domaine de la sonorisation, ils égalent et dépassent ceux obtenus par des systèmes plus compliqués et plus onéreux.

Dans le domaine de l'écoute (de contrôle ou de salon) sans prétendre à l'hyper-fidélité, on obtient une audition de qualité sans encombrement exagéré et par des moyens relativement simples.

Nous ne pouvons prévoir quel sera l'avenir des réflecteurs sonores, mais on peut être assuré, d'ores et déjà, que l'on ne devra plus les négliger.

Discussion

FLORISSON, C.: En quoi est faite la conque?

CONFÉRENCIER: Les conques sont actuellement construites en staff, mais une fabrication en matière plastique est actuellement à l'étude.

SMIT, H. J. A.: Quelles sont les fréquences de résonance basse du haut-parleur utilisé et de la partie inférieure du pavillon?

CONFÉRENCIER: La fréquence de résonance de la membrane du haut-parleur est de 80 Hz environ, tandis que la fréquence de résonance du pavillon est de l'ordre de 40 Hz.

THE APPLICATION OF THE HAAS EFFECT TO SPEECH REINFORCEMENT SYSTEMS

by P. H. Parkin

Presented by A. T. Pickles

Building Research Station, Garston, Watford, England

Summary. The paper points out that, although it had long been obvious that speech re-inforcement systems using several distributed loudspeakers could be improved by arranging for the sound from the various loudspeakers to arrive more or less simultaneously at the listeners, it was not until the work of Haas that it was known how to adjust the successively arriving sounds in amplitude and in time sequence so that maximum realism and intelligibility were achieved. Three recent systems in Great Britain which apply the Haas effect are described, the most important one being that in St. Paul's Cathedral, London. This last system also employs directional loudspeaker arrays (columns).

Sommaire. On savait depuis longtemps que les systèmes amplificateurs acoustiques comprenant plusieurs haut-parleurs répartis dans une salle pouvaient donner des résultats plus satisfaisants si on s'arrangeait pour que les sons émis par ces haut-parleurs arrivent aux auditeurs plus ou moins simultanément; mais, comme il est rappelé dans le présent article, il a fallu attendre les travaux de Haas pour connaître les amplitudes, ordre de succession et retards à donner à ces sons pour que l'intelligibilité et la sensation de réel soient optima. On décrit 3 systèmes réalisés récemment en Angleterre et dans lesquels on utilise cet effet Haas. L'installation la plus importante est celle de la Cathédrale St. Paul, à Londres; elle comporte aussi des séries de haut-parleurs directifs, disposés en colonnes.

Zusammenfassung. Obwohl seit langem bekannt ist, daß Verstärkeranlagen mit mehreren verteilten Lautsprechern verbessert werden können, wenn der Schall aus den verschiedenen Lautsprechern mehr oder weniger gleichzeitig beim Hörer eintrifft, weiß man doch erst seit der Arbeit von Haas, wie man die nacheinander eintreffenden Schallsignale nach Amplitude und Zeitverzögerung einzustellen hat, um maximale Wiedergabetreue und Verständlichkeit zu erreichen. Es werden drei neuere Anlagen aus England, die den Haas-Effekt ausnutzen, beschrieben, von denen die wichtigste die in der St. Paul's Kathedrale in London ist. Bei der letzteren Anlage werden gerichtete Lautsprecherzeilen verwendet.

It has long been recognised that the intelligibility of speech re-inforcement systems using multiple loudspeakers could be improved by arranging for the sound from the various loudspeakers to arrive at the listeners more or less simultaneously. The recent work of Haas [1] has provided some quantitative results for design purposes and has also demonstrated a hitherto unexpected fact, namely that when a single echo is delayed behind the original sound by between 5 and 25 milliseconds, this echo has to be 10 dB greater in intensity than the original sound to be heard equally loud (Fig. 2 in ref. [1]). Subtracting an arbitrary 3 dB, we can assume that an echo may be up to 7 dB greater in intensity (approximately double in loudness) and not be detectable as a separate source. That is to say, the apparent direction of the sound will be determined entirely by the original source. The "Haas effect" has obvious implications in the design of speech re-inforcement systems, and this paper describes three recent systems in Great Britain which use this effect to maintain realism (i.e. to maintain the apparent direction of the source) and to improve intelligibility.

The first system is that used in the new *Free Trade Hall*, Manchester. Here the main loudspeakers are mounted in the front edge of the orchestral canopy above the platform but are out of sight of the audience seated under the balcony. In this area the speech is supplemented by loudspeakers mounted on the balcony soffit and which are delayed by intervals corresponding to the path differences from the platform to these areas, plus 5 to 25 ms. Their amplitudes are adjusted so that the sounds from them are not more than 7 dB greater than that arriving from the platform, and thus all the sound appears to come from the platform. Now the sounds from these supplementary loudspeakers travel back towards the platform, and, aggravated by the added delays, might be so far behind the original sounds as to be disturbing. A further Haas result shows the degree of disturbance to be expected from echoes of intensities from + 10 dB to − 10 dB arriving at time delays of up to 160 ms (Fig. 4 in ref. [1]). This result enables us to decide on the necessary directionality of the supplementary loudspeakers in order to prevent such disturbance; in this case it was 6 dB (front-to-back ratio).

The time delay mechanism [1]) in this Hall consists of a rotating tufnol disc coated on the edge with a magnetic material. The heads are spaced 0.025 mm away from the edge and there is therefore no wear.

[1]) Electric and Musical Industries Ltd.

A more elaborate system is that installed in *St. Paul's Cathedral* [2]. Here the problem was to reinforce speech from the altar, the pulpit and the lectern; the horizontal distance to be covered, in the case of the altar, was about 110 m, under very reverberant conditions. At each of three speech positions a loudspeaker column 3.3 m long was used, supplemented by shorter columns down the nave. Thus when the pulpit (or lectern) is used, the pulpit column is in circuit and the first pair of nave loudspeakers are delayed and their amplitude adjusted so that the sound from the pulpit arrives first (by 5 ms) and so that the intensity of the sound from the nave loudspeakers is 7 dB greater than the sound from the pulpit column. The second and third pairs of nave loudspeakers are similarly delayed (by 10 and 15 ms respectively) but it was found that the reference intensity could be taken as that arriving from the first pair of nave loudspeakers. Thus greater gain was possible, and because for a listener at the rear of the nave the first pair of nave loudspeakers is in the same direction as the pulpit, the visual influence is sufficient to give the impression that all the sound is coming from the pulpit. For speech from the altar, the altar column is the prime source and the pulpit and lectern columns are both on at appropriate amplitudes and suitably delayed; the delays on the nave columns are also adjusted suitably, all these operations taking place automatically when the altar microphone is switched on by the operator.

It might be mentioned that columns which were long enough to be sufficiently directional at the lower frequencies were too directional at the high frequencies. Several types of column were tried in order to overcome this difficulty; the most satisfactory arrangement was found to be to use two parallel lines of loudspeakers in each column, one line being the full length of the column and the other line being one quarter the length, with a cross-over network operating at 1000 c/s. This arrangement had the added advantage that the small loudspeakers (8 cm diameter) used for the high-frequency line were not significantly directional in the horizontal plane.

Although the main purpose of the time-delays in this cathedral was to maintain realism, the intelligibility was considerably improved. At the rear end of the nave the word-intelligibility in the empty cathedral (reverberation time 11½ s at 500 c/s) was 85% with delays, compared with 52% when undelayed.

The third system is that installed in *Harringay Arena*, London. This arena is oval in plan and is used for a variety of purposes. Two apparent sources of sound are required, the first being at one end of the arena (e.g. when used for ice-shows) and the second being at the centre (e.g. when used for boxing matches). Some 60 loudspeakers are suspended above the audience at the points of intersection of two sets of concentric circles centred at the two apparent sources, and the time delays and amplitudes are adjusted depending on the source to be used. The transition from one source to the other can be made in ten minutes.

The delay mechanism [1]) used in *St. Paul's Cathedral* and in *Harringay Arena* consists of a rotating disk of plastic magnetic material extending beyond the periphery of a supporting turntable. The heads are in contact with this annulus, and when the disk becomes worn (the minimum life is 500 hours) it can be replaced in two minutes at the cost of a few shillings.

All these systems have been designed on the basis of the original HAAS results, and have proved satisfactory in use. More recent work (as yet unpublished) using multiple echoes may modify slightly the original results, but probably only to make the design criteria less critical.

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References

[1] HAAS, H., Acustica 1 [1951], 49. (In English: Building Research Station Library Communication No. 363, [1949].)
[2] PARKIN, P. H. and TAYLOR, P. H., Wireless World 58 [1952], 54 (2) and 109 (3).

Discussion

GAYFORD, M. L.: Did you encounter any trouble in the magnetic disc delay unit due to head wear, as the magnetic material is in contact with the faces of the heads.

LECTURER: The pressure of the material on the heads was very slight, and no trouble has been caused by wear on the heads although, as mentioned, regular replacement of the magnetic discs is needed.

HUNT, F. V.: It would be interesting to compare the sound decay curves obtained with the source at the normal position of the speaker, and with the re-inforcement system first off and then on. Differences in the shape of the initial portions of the decay curve should be of assistance in the interpretation of curves obtained in smaller rooms.

JORDAN, V. L.: With a delay system the start of the decay period would show an abrupt fall after the arrival of the direct and time-delay sounds, while with an un-delayed system there would be more steps in the falling-off of the reverberation curve.

The Lecturer agreed with Dr. JORDAN, and regretted that it had not so far been possible to take pulse photographs in any of the auditoria decribed.

BEGUN, S. J.: What is the reverberation time of a room in which a time-delay system is installed?

LECTURER: In view of the comparative length of any reverberation time compared with the milli-second changes due to a time-delay system, one would not expect any appreciable change in the reverberation time averaged in the usual way over a decay of at least 30 dB.

CREMER, L.: In a new concert-hall in Munich I have installed

[1]) Pamphonic Reproducers Ltd.

in the ceiling two groups of loudspeakers for speech reinforcement, with time delays of different magnitude, over the rear stalls and over the balcony. An echo could be heard near the platform if the amplification was not adjusted to the correct loudness-level. After installing a third group in the ceiling near the stage working without artificial but with this small natural time-delay the echo was suppressed even at some higher loudness-level.

LECTURER: The original HAAS results, based on a single echo, will probably be modified a little when more than one echo is present, and I hope that Professor MEYER will soon present the complete facts.

SOUND-REFLECTING SURFACES

by F. W. J. SCHWEIGMAN
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Summary. The human voice of average power can be well heard and understood by an audience larger than usually taken for granted, if the sound energy is well utilized by reflection of walls, ceiling and partitions. It is possible to design the reflecting surfaces in such a manner that for all listeners the sound level and intelligibility have the same value, independent of the varying distances from the speakers.

Sommaire. La parole humaine de puissance moyenne peut être entendue et bien comprise par un auditoire plus grand que celui admis généralement si on utilise toute l'énergie sonore grâce à la réflexion sur des murs, plafonds, planchers et cloisons. Il est possible de prévoir les surfaces réfléchissantes de façon que le niveau sonore et l'intelligibilité soient les mêmes pour tous les auditeurs, quelle que soit leur distance au conférencier.

Zusammenfassung. Menschliche Sprache von normaler Lautstärke kann in einem Saal viel weiter gehört und verstanden werden, als man gemeinhin annimmt, wenn die Schallenergie durch Reflexion an den Wänden, an der Decke und an Zwischenwänden nutzbar gemacht wird. Es ist möglich, die reflektierenden Flächen so zu konstruieren, daß für alle Hörer unabhängig von ihrem Abstand vom Sprecher Schallpegel und Verständlichkeit gleich sind.

The most simple case of application of sound-reflecting partitions arises when there is only one source of sound, located in a fixed, unvariable place. This situation occurs e.g. in an auditorium with a cathedra, or in a church with a pulpit.

Let us suppose that each sound wave, or rather sound ray, after striking a surface, is reflected towards the audience. Let us also assume that this audience is perfectly absorptive, and that there is no place in the audience where the reflected sound arrives more than 1/16 s after the arrival of the direct sound. Then all emitted sound energy would be received by the audience without influencing or disturbing the audibility or the intelligibility by reverberation.

In each point of the audience the sound intensity I consists of two parts, the direct and the reflected sound. Thus:

$$I = I_{dir} + I_{refl}.$$

Should it be possible to design the surfaces so as to make I constant for all points of the audience, then the distribution of the sound energy would be most favourable. The sound level would be equal for all listeners, and therefore the intelligibility too, at least if noise level and rest-reverberation are also equal for every listener.

First of all we shall calculate the value of this sound level in a space of given dimensions, with a source of sound of a given power; next we shall examine how large these dimensions may be that the audibility of the source is still sufficient.

This source may be the human voice with a power of 35 μW; we assume that the audience fills a semi-circular space, the source being placed in the centre. The source is supposed to be elevated to a height a above the horizontal plane of the audience. For this horizontal plane we choose polar coordinates p and β, the centre being the vertical projection O′ of the source O. Using spherical coordinates for the fixation of the direction of an emitted sound ray and denoting the power emitted per unit of solid angle in a direction θ, φ by E, we obtain

$$\int_0^{2\pi} \int_0^{\pi} E \sin\theta \, d\theta \, d\varphi = 35 \times 10^{-6} \text{ W}.$$

If the sound intensity I is the same for all points of the audience, then

$$I \int_{\beta_{min}}^{\beta_{max}} \int_{p_1}^{p_2} \frac{a}{\sqrt{p^2 + a^2}} \, p \, dp \, d\beta = 35 \times 10^{-6} \text{ W}.$$

Now for our simple shape of auditorium $p_2 = p_{max}$ = constant, and $\beta_{max} - \beta_{min} = \pi$; therefore the equation becomes after integration

$$I = \frac{35 \times 10^{-6}}{\pi \, a \, (\sqrt{p_2^2 + a^2} - \sqrt{p_1^2 + a^2})}.$$

From this we can calculate now the value of the constant intensity I in its dependence on the maximum distance p_2; the result is as shown in the Table.

TABLE

The intensity, sovnd level ond number of persons seated as a function of p_2; for comparison the equivalent open air distance p_e.

p_2	I	*sound level*	*number of persons*	p_e
m	W/m	dB		m
20	40.5×10^{-8}	56.1	750	4.7
40	16.5	52.2	3000	7.6
60	10.4	50.2	6700	10.0

We have calculated the number of auditors in a full house, assuming that this number equals 1.2 × the gross area in square meters. a has been put equal to 2 m, p_1 to 6 m.

The computed values of the sound pressure level seem to be rather low. We must not forget, however, that the mere sound level is not a standard for the intelligibility, and that we assumed that the emitted sound rays, due to one and the same sound impulse, do not arrive at a point of the audience with intervals larger than 1/16 s. In this favourable condition the sound level needs not to be 60 dB or even more, but a lower value will do. This can be made clear by introducing the equivalent open air distance p_e (last column of the Table). p_e is defined as the distance in the open air (i.e. no reflections at all) for which the intensity at the listener equals the intensity previously found (reflectors being present; see column 2 of the Table). It can be computed from the well-known formula for the direct sound: $I = E/(p_e^2 + a^2)$. E can be obtained from the total power of the source and the well-known directivity characteristics of the human voice.

E.g., a sound level of 52.2 dB (3rd column) is not high, but the listening conditions being very favourable indeed, the situation is undoubtedly comparable with the open air situation of the same intensity, i.e. at $p_e = 7.6$ m, and at this distance the intelligibility is certainly not insufficient.

However, we may not give too much weight to the numbers given in the table, for we have left too many factors out of consideration. In the first place we assumed that the total power output of the source could be utilized without any loss; but this is impossible. There will always be a certain amount of energy, which cannot be used as planned. Then the atmospheric absorption has not been taken into account; this absorption may not be neglected for the higher frequencies and for distances somewhat over 15 m.

Two more points were left out of consideration in our calculation. We started from the distribution of the *total* energy output over the different directions. But this average distribution has no real meaning, for in reality the distribution depends strongly on the frequency. Therefore the distribution of the energy over the frequency range should be taken into account, as well as the contribution of the frequencies to the intelligibility. Lastly we have neglected the attenuation of the sound, travelling over the highly absorptive surface of the audience to the rear of the auditorium.

Only the first two factors (inevitable loss of energy, and atmospheric absorption) will cause a decrease of the maximum value obtainable of the constant sound level. The other factors have an influence on the design of the reflecting partitions.

For all the above mentioned reasons the numbers of the Table can give at best a qualitative picture, but do not give exact values of the actual sound level. But even if the necessary corrections were made, yet the calculated value of the sound level would not be a reliable standard for the intelligibility, for the inevitable noise level will reduce this intelligibility.

The above mentioned equivalent distance may be of use in this respect however. For it is known by experience at what distance the auditor can hear sufficiently well the speaker facing this listener, and speaking with an average power output of 35 μW, the noise level being 35 dB. If both speaker and listener are located in the open air on a highly absorptive field so as to avoid increase of the audibility by reflected sound, then at a distance of 9 m the listener will be able to hear the speech without effort. Now we can calculate the maximum audience for which the constant sound level equals the value of the open air sound level at a distance of 9 m from the speaker.

If the power emitted within a solid angle of $0.6 \times 4\pi$ steradians can be utilized (the power in the remaining angle $0.4 \times 4\pi$ steradians being lost), the atmospheric absorption being taken into account by a loss of 0.2 dB per 10 m, then the result is that the maximum audience amounts to 2700 listeners.

That is why we maintain that for an audience of less than 2700 persons the human voice need not be amplified, if at a distance of 9 m in front of the source the speech can be heard well. This conclusion assumes the possibility of obtaining a constant sound pressure level by means of reflecting partitions and it must now be examined whether this indeed is possible. So we have to answer this question: Can a possible design of walls, reflecting partitions and the like have the result that for each place in the audience the sound level, as well as the intelligibility, has the same value?

For the design of the shape of these surfaces we choose spherical co-ordinates r, θ and φ, the source

of sound being at the centre. Should there be an answer in the affirmative to this question, then it must be possible to have the equation of the surface $r = f(\theta, \varphi)$ derived from the following condition of constant intelligibility:

$$\frac{E_{dir}}{p^2 + a^2} + \frac{E_{refl} \sin\theta \, d\theta \, d\varphi}{\frac{a}{\sqrt{p^2 + a^2}} p \, dp \, d\beta} = \text{constant.} \qquad (2)$$

Now E_{dir} depends on p and β for several reasons: the emitted power varies with the direction from the source; the attenuation due to atmospheric absorption depends on the distance p, as well as the attenuation caused by the absorption of that part of the audience for which the distance from the source is smaller than p. In the same way E_{refl} depends on θ, φ, p and β. Therefore eq. (2) can be written

$$F(p, \beta, \theta, \varphi) = 0. \qquad (3)$$

The sound, emitted into the direction θ, φ from the source, will strike the wall or reflecting partition at the point r, θ, φ. This sound is reflected towards the point p, β of the audience; therefore the position of the tangent plane at the point r, θ, φ is determined by these five co-ordinates.

This position of the tangent plane is also determined by the partial derivatives of r.

Hence $$\frac{\partial r}{\partial \theta} = f_1(r, \theta, \varphi, p, \beta) \qquad (4)$$

and $$\frac{\partial r}{\partial \varphi} = f_2(r, \theta, \varphi, p, \beta). \qquad (5)$$

Thus we have three independent equations, and three unknowns, namely p, β and r. Therefore it is theoretically possible to determine r as dependent of the independent variables θ and φ.

It lies beyond the scope of this paper to go into the details of the practice of these calculations. This method of calculation has been applied to the design of sound-reflectors above the pulpit in churches.

SECTION III

ACOUSTIC MEASUREMENTS

GENERAL REVIEW

by L. L. BERANEK
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1. Introduction

In his opening address, Professor BOLT ably summarized the early work in electro-acoustics, and pointed out the close relation between bio-acoustics and physical acoustics. It will not be necessary, therefore, for me to repeat the historical material. I shall confine my review to developments during the current decade.

Acoustical measurement techniques have been advanced, during this decade, to yield greater precision and to cover a wider range of applications than ever before. Measurements on both physical and psychological quantities are performed in many laboratories of acoustic research. We are able to specify with considerable accuracy the properties of a sound wave, not only in terms of its intensity but also in terms of its spectrum, its wave form, its phase characteristics and, in some cases, the response it evokes from an average listener.

Our most rapid progress in the field of acoustic measurements was made after the miniature condenser microphone became available on a widespread basis. With a stable microphone of this type, accurately calibrated as a function of frequency, we were able to discuss with confidence the properties of microphones and loudspeakers, and to specify reliably the stimulus delivered to the listener.

Our next important advance in measurement was in making widely available filter sets with band widths of an octave, half-octave or third-octave. Such filters are now widely used for analyzing the behaviour of sound in buildings and for specifying the spectra of noise fields.

With accurately calibrated microphones and analyzers we can determine the power output, the frequency spectrum and directivity characteristics of noise sources. A quantitative specification of these noise source characteristics gives us essential information for acoustical engineering design. We can now proceed on a solid physical basis, taking into account all of the energy that is produced by a noise source in a bounded system. We can readily understand, in terms of total power, why it is enormously simpler to quiet the spoken voice which has a power output of about 10^{-5} W, than to quiet a jet aircraft engine with a power output of 10^{4} W.

Even within the past five years it has become easier to determine, in a routine engineering analysis, the wave form and time pattern of a complex sound. This we do now with a cathode ray tube fitted with a camera that produces a developed picture within a minute after the event.

Suppliers of high quality equipment have in recent years brought out more reliable graphic level recorders, improved vibration pickups, smaller microphones, versatile electronic noise sources, and associated oscillators, amplifiers and calibrators. Research laboratories can therefore devote much more of their effort to acoustic studies *per se*, without diluting their work with routine construction of equipment.

In many types of electro-acoustical engineering problems we must know as much about the listener as we do about the source. As a matter of economy, one should not over-design in noise control or in refinement of an auditorium. One should match the noise condition or the acoustic design of a room as

closely as possible to the dictates of the observers and their ears.

This practical need has stimulated the interest of workers in psycho-acoustics, and they, in turn, using improved measurement techniques and equipment, have begun to establish damage-risk levels for the ear, speech interference levels, permissible echo levels, permissible reverberation times, neighbourhood annoyance levels and other criteria for acoustic design.

With a knowledge of the characteristics of the source and the tolerance limits of the listener, we can obtain directly the amounts by which the noise or the reverberation must be reduced. Here again, recently obtained information on the acoustic behaviour of building structures, mufflers and atmospheric attenuation assists us in designing structures that provide required noise or echo reductions at reasonable cost.

At this great Congress, during the coming days, we may expect to receive new information that will advance our ability to measure and to evaluate acoustic problems on a wider scale. In this introductory survey for Section III, I shall make some comments on technical developments in acoustic measurements, and also suggest some of the directions in which I believe major research efforts should be pointed in the near future.

The most difficult measurements are those involving the reaction of human beings to physical stimuli. Because of the difficulties inherent in this type of experimentation, determination of basic psycho-acoustic function has lagged behind physical measurements and constitutes one of our most urgent needs. Accordingly, the first part of this paper is devoted to some of the problems that exist in the psycho-acoustic field. After that we shall discuss, in order, noise measurement, acoustic impedance measurement and mechanical impedance measurement.

2. Psycho-acoustic measurements

Let us start at the basic level with a discussion of audiometric measurements. A paper on this subject appears in Section IV of these meetings. Currently, the standard audiometric threshold of the United States is higher than the proposed standard threshold of England by about 10 dB. The differences are hard to explain but there is every reason to believe that they depend upon measuring techniques, and not upon national differences in the population.

The American threshold curve is the result of national hearing surveys performed on subjects who volunteered for tests. The British data were taken on selected subjects both at the National Physical Laboratory and at the R.A.F. Central Medical Establishment. It is also well known that laboratory tests in the United States of America yield a threshold curve more in conformance with the proposed British curve than did the U.S. Public Health and World's Fair Surveys, just mentioned.

Difficulties very likely arose in the American surveys from the type of people who volunteered for tests. It may be that people with some hearing loss are more likely to volunteer than those with no hearing loss, and therefore that subjects selected on a random sampling basis (from the general population) would make for more significant tests.

The background noise levels at the time of measurement also affected the results of the measurements in the two countries. Indeed, it seems to me to be a necessary part of every audiometric experiment to produce an identical, reproducible and fully specifiable noise background at the ears of the subject. It would seem to be unreasonable to say that the low British threshold is the "right" one if to obtain it hearing loss must be measured under abnormally quiet conditions that do not exist in the best concert halls, theatres, offices or average homes. A survey of background noise conditions in such locations might be made and used in establishing a standard noise background for audiometric measurements.

Of equal importance in assessing the difference between the British and American thresholds is the behaviour of the earphone on a listener's ear. Anyone who has measured sound pressure levels in the ear canal as produced by a particular earphone knows that the values fluctuate by 5 to 20 dB from person to person at both high and low frequencies due to the size of the pinna, the size of the ear canal and the tightness of the seal against the head. This variation can be reduced by proper choice of earphone cushion and by using a driving unit that has low acoustic impedance. However, it seems reasonable that the most effective solution would be to attach a flexible probe tube microphone permanently to the earphone that would sample the sound field in the outer ear canal and would give the experimenter a more precise knowledge of the physical stimulus.

We ought now to re-examine our position and to adopt a more nearly correct international standard threshold curve for use in audiometric measurements and as a base for equal-loudness contours.

The question of a loudness scale in sones as a function of loudness level in phons also needs re-examination. It would seem to the average scientist that, given a competent psycho-acoustician to perform te task, it would be a straightforward endeavour to determine a loudness function for a 1000 c/s pure tone, Recently, however, a clash has

occurred between the original results of FLETCHER, STEVENS et al. and some new results of GARNER in the U.S.A. Two loudness scales now exist, purporting to be the same thing, that at first glance hardly seem to resemble each other.

Part of the difference seems to arise from a well-known difficulty. On the one hand, the subject in a listening experiment must not be told too much about how he is to react to the stimulus because he may only parrot the answer that the experimenter wants. On the other hand, the experimenter must so instruct and interrogate the subject that he knows that the judgement rendered is pertinent. Let us look at this a little more closely.

Basically there are at least two techniques that are used for determining a loudness function. One is to ask the listener to increase or to decrease the loudness of a tone of a given frequency by factors of two, four, or ten. The other is to ask the listener to divide the interval between two tones of the same frequency, but of different intensities, into two to four parts of equal incremental loudness. The technique of equal ratios of loudness was used by FLETCHER and STEVENS et al. and their colleagues, while the technique of equal increments of loudness was used by GARNER. The more sound technique is believed to be that used by GARNER.

GARNER's subjects might possibly be confused between equal interval judgements and equal ratio judgements so that neither type of scale emerges in the averaged results, but rather something in between. It is also possible that his results might be obtainable from the equal ratio experiments if the subjects were properly instructed.

It is interesting to note that both loudness scales reliably permit determination of the *loudness level* of a groop of tones, given the loudness levels of the component tones.

There are other branches of psycho-acoustics in which better subjective measuring techniques are needed. In the American literature there will soon appear tentative information on damage risk to hearing and on the annoyance of certain sounds. These are matters of international interest. We are also greatly interested in such subjects as criteria for musical quality in sound systems and criteria for the design of auditoriums and listening rooms. We urgently need better techniques for measuring subjective responses to stimuli involving noise, music and other complex sounds. In Section I of this Congress we shall hear a paper on the subjective assessment of quality standards, which should be of interest to many of us.

It might be pointed out that studies of subjective factors are particularly difficult because the human mechanism does not obey the laws of a linear system in which superposition holds. For example, one may set up a mathematical model for explaining the ability of an individual to detect the presence of a pure tone in a background of random, white noise. This model leads to the so-called "critical bands" theory of hearing. However, when other types of stimuli are applied to the ear, this mathematical model fails, as for instance, in cases where transient sounds or complex sounds are involved.

Perhaps inclusion of this discussion of psycho-acoustic measurements in this review may encourage a few young scientists to work in this field. The discussion might also be useful as supporting evidence in a request for funds to support this type of research. The field of psycho-acoustics is not one in which either the physical scientist or the psychologist can work alone. Close co-operation between physical and psychological scientists is necessary if effective advances are to be made in reasonably short periods of time.

3. Measurement of sound pressure levels

Second only in importance to the previous topic is the necessity for microphones that will measure airborne sounds:

1. with frequencies up to 50 000 c/s;
2. with sound pressure levels up to 200 dB;
3. with wind velocities up to 100 m/s;
4. with ambient temperatures up to 500 °C;
5. with ambient pressures from as low as one tenth of an atmosphere to as high as 10 atmospheres.

In regard to the first item above, we note that the response of present microphones is generally quite low at 50 000 c/s. This results from one of two situations: Either the microphone has good sensitivity, and the resonance frequency f_0 of the microphone diaphragm is low so that response decreases above f_0 by 12 dB per octave; or the microphone diaphragm is made very stiff so as to raise the resonance frequency, in which case the sensitivity is also very low.

A possible solution to a better high frequency efficiency is to develop, by means of active networks, a microphone with a diaphragm whose acoustic impedance as viewed from a plane wave is as near the characteristic impedance ϱc of air as possible. In this manner, the response can be held constant with frequency and the output can be made high.

As to the second item, sound pressure levels as high as 200 dB can now be measured with currently available microphones. These microphones are of the condenser type and appear to be more rugged than earlier types, although the stability of their calibrations over a wide range of ambient temperatures and ambient pressures has yet to be determined.

In regard to item 3, sounds must often be measured outdoors under conditions of moderate and gusty winds. Although many types of windscreens have been shown in the literature, almost no systematic investigations of the sources of noise around microphones and the factors of scaling in the design of windscreens have been reported before now. In the first paper of Section III this afternoon, we shall learn the results of one systematic study involving spherical and cylindrical screens around pressure microphones.

At higher steady wind speeds, some tentative studies have shown that a section of symmetrical aerodynamic foil, or a teardrop construction with the microphone located at the blunt end at the point of stagnation, leads to the best results. In this case, a sintered metal cover over the microphone, moulded to preserve the shape of the enclosure appeared to give the best results. Further research on this type of microphone enclosure for high-speed steady wind streams is badly needed as there are not even qualitative data on their performance above 50 m/s.

To measure sound in the last two catagories, microphones for operation at high temperatures and with high and low ambient pressures are necessary; these are used for investigation of the noise in engine cylinders, in exhaust systems, and in engine test chambers. Here microphones must operate at temperatures up to say 500 °C and pressures between 0.1 and 10 atmospheres. Probably this is not a difficult design task with such materials available as stainless steel and other alloys. Nevertheless, the need exists for microphones that will reliably measure sound pressures in such widely varying conditions of operation.

4. Measurement of acoustic impedance

Much has been written on the subject of acoustic impedance measurement, yet very few laboratories report results because of the difficulties of taking data and of reducing them to the desired form. The three most common impedance measuring techniques involve probing a standing wave, measuring the complex pressure in front of the sample for two known impedances as well as for the unknown impedance so as to eliminate the source effects. The standing wave method involves a long tube which is large and slow to use and with which the necessary calculations are quite tedious. The short-tube method requires accurate measurement of the sound pressure in amplitude and phase before and after adding the sample, and it involves assumptions about the constancy of output of the acoustic source used. The comparison-impedance type of measurement eliminates the source, but it results in equations that are difficult to manipulate. In other words, the present methods are cumbersome to use or are too approximate to make measurement of acoustic impedance a usual, rather than an unusual, procedure.

During these meetings we shall hear papers in Section III from the Netherlands, Great Britain, Denmark and the U.S.A. on the short-tube and the comparison-impedance methods of measuring acoustic impedance. These papers will probably shed light on the road toward continuing research.

There is justification for the opinion that we shall not find widespread use for acoustic impedance measuring equipment until we develop a microphone that will measure simultaneously, as near a point as possible, the sound pressure and the particle velocity. The measurement of these quantities is not an unusual matter, except that stability of both amplitude and phase over a wide frequency range for both particle velocity and sound pressure has not been achieved. Ribbon microphones for measuring particle velocity do not possess phase stability, as was demonstrated by FIRESTONE and CLAPP over a decade ago. Pressure microphones of the condenser type are stable if properly designed. Perhaps the answer to a stable microphone for measuring particle velocity is to use a condenser microphone of the pressure gradient type such as was developed in Germany some years ago.

If stable microphones for measuring pressure and particle velocity are available, we need only a circuit to take the ratio of pressure to particle velocity and a phase meter to measure differences of phase angles in order to determine acoustic impedance.

With the same transducers, the product of pressure and particle velocity could also be measured and from it and the phase angle would come the acoustic intensity and the reactive power per unit area.

5. Measurement of mechanical impedance

Noise is often produced by vibration of mechanical devices or parts of such devices. Before we can reduce this noise, we should measure the vibration characteristics of the device and then we should prescribe corrective measures on the basis of the data. To make the measurements, we need suitable instruments.

Modern vibration meters have been in use for the past 25 years. They measure, in general, the vibration displacement, velocity or acceleration. The results obtained with these instruments are, however, of only limited value.

A relatively undeveloped type of measuring instrument is one for determining the ratio of force to velocity at a point in a mechanical system. An increasing number of noise problems require a high

degree of reduction of mechanical vibration. One way of decreasing mechanical vibration is by trial and error using vibration isolators and damping materials wherever deemed necessary. Another, and the more basic way, is to determine separately the vibration characteristics of the source, the transmitting path and the receiving mechanism, and to deduce logically from this information the most economical ways for reducing the vibration at the ultimate point of reception.

For example, a different type of vibration isolating mount is necessary beneath a motor to reduce the vibration transmitted into a supporting structure by 20 dB if the motor stands at a point of low mechanical impedance than if it stands at a point of high mechanical impedance.

Just as we need a simple, high-speed method for measuring acoustic impedance, we must also develop a simple, high-speed method for measuring mechanical impedance. There is also a need for force measuring devices that can be inserted easily between parts of a mechanical system.

6. Concluding remarks

In the previous sections of this paper four branches of electro-acoustics have been mentioned: psychoacoustics, physical acoustics, and instrumentation for measuring sound and mechanical vibrations. In each of these branches lie rich opportunities for research. We may look forward to many more congresses before the possibilities of new advances have been exhausted.

NOISE IN CONDENSER MICROPHONES

by A. G. Th. Becking and A. Rademakers
Philips Research Laboratories, Eindhoven, Netherlands

Summary. It is pointed out that the mechanical resistance, used for damping a simple mechanical oscillator, acts as the source of a fluctuating force with spectral intensity $4RkT$. In the case of a condenser microphone this results in a noise pressure on the diaphragm, which lies in the region of audible sound pressures. Experimental results, obtained with two microphones, agree well with the theory.

Sommaire. La résistance mécanique qu'on emploie pour amortir un oscillateur mécanique se comporte comme une source de force fluctuante ayant une intensité spectrale de $4RkT$. Cela veut dire que, dans le cas d'un microphone condensateur, il s'exerce sur la membrane une pression de bruit qui est de l'ordre de grandeur des pressions sonores audibles. Les résultats expérimentaux obtenus avec deux microphones cadrent bien avec les résultats théoriques.

Zusammenfassung. Der mechanische Widerstand, den man für die Dämpfung eines mechanischen Oszillators braucht, erscheint als Quelle einer Rauschkraft mit der spektralen Intensität $4RkT$. Im Falle eines Kondensatormikrophons bedeutet dies einen Rauschdruck auf die Membran, der im Bereich der hörbaren Schalldrücke liegt. Ein experimenteller Nachweis an zwei Mikrophonen ergibt gute Resultate.

In general the combination of a microphone and an amplifier shows a certain output noise voltage, setting a lower limit to the sound pressures which can be discerned by the microphone.

An important part is played in this respect by the input stage of the amplifier, but the microphone itself acts also as a source of noise. In the case of electrodynamic microphones for instance, with a sensitivity of about $1.5 \cdot 10^{-2}$ V/Nm^{-2} (1.5 mV/μbar) at a resistance of 10 000 Ω, the r.m.s. noise voltage of the microphone, calculated from the resistance value, is about $1.5 \cdot 10^{-6}$ V, or 60 dB below the output voltage at 0.1 N/m^2 (1 μbar). In practice this is of the same order as the noise of the input stage of the amplifier.

In the case of condenser microphones the problem is more complicated. The microphone at zero sound pressure is often considered as a fixed condenser, only influencing the noise of the input stage by its capacity. Yet a condenser microphone has a self-noise, and in the following we shall show that it is of the same order of magnitude as that of the above mentioned electrodynamic microphone.

In order to point out where this self-noise originates from, we will consider the behaviour of the membrane somewhat closer. Below and in the neighbourhood of the first resonance frequency it may be looked at as a simple harmonic oscillator, with a suitable damping applied to it, in order to eliminate the resonance peak. Only a small part of this damping can be given by the radiation resistance. Most of it must be introduced artificially and this is normally done by taking advantage of the motion of the air behind the diaphragm (Fig. 1). As this moves to and from the electrode, part of the air in the narrow gap must flow to and from the slots

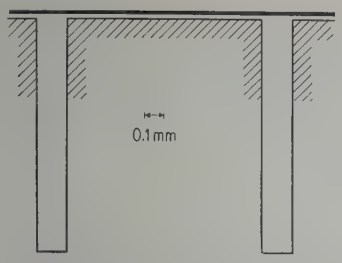
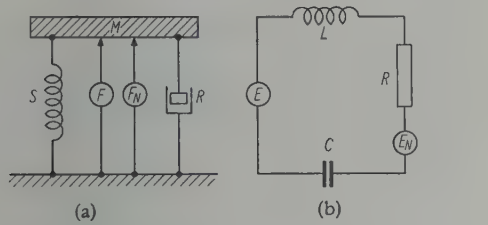


Fig. 1. Cross-section through the microphone.

or holes and because of the narrowness of the gap this motion gives rise to a friction which can be made sufficient to smooth out the resonance peak.

It can be shown now that this mechanical resistance is the source of a noise force just as an electrical resistance is the source of a noise voltage. We will illustrate this in Fig. 2 which shows a well-known electromechanical analogy. Fig. 2a gives a schematic diagram of our damped mechanical oscillator, Fig. 2b a damped electrical oscillator.



(a) (b)

$$M\ddot{x} + R\dot{x} + Sx = F + F_N \qquad L\ddot{Q} + R\dot{Q} + \frac{1}{C}Q = E + E_N$$

$$\overline{F^2_N}\Big|_{\nu}^{\nu+d\nu} = 4RkT\,d\nu \qquad \overline{E^2_N}\Big|_{\nu}^{\nu+d\nu} = 4RkT\,d\nu$$

Fig. 2. Diagrams and differential equations for simple mechanical and electrical oscillators.

The differential equations for the amplitude of the diaphragm and for the charge of the condenser are the same. In the electrical case it is well-known that the theory of fluctuations leads to a noise voltage, with spectral intensity $4RkT$ in series with the resistance R, so that the differential equation must be completed with the last term E_N. The electromechanical analogy now requires the introduction of an extra term of the same form in the mechanical equation: a fluctuating force F_N with spectral intensity $4RkT$ appears in parallel to the external force F. In the case of the microphone this fluctuating force means a noise pressure on the diaphragm, with a flat frequency characteristic, and this will produce an electrical noise output with the same spectral distribution as the normal frequency characteristic of the microphone.

To give an idea of the order of magnitude of this pressure we will consider a high quality condenser microphone as used in the Philips Laboratories. The diaphragm has a mass $M = 10^{-5}$ kg (10 mg) and an area $A = 1.8 \cdot 10^{-4}$ m² (1.8 cm²). The resonance frequency is laid at $\nu_0 = 14\,000$ c/s, which means a stiffness $S = 8 \cdot 10^4$ N/m. The damping is nearly critical, hence

$$R = 1.4\,S/2\pi\nu_0 = 1.3\ \mathrm{N/ms^{-1}}.$$

From this value of R we find for the r.m.s. noise pressure over the frequency range $0 \cdots 14\,000$ c/s:

$$p_n = 1.0 \cdot 10^{-4}\ \mathrm{N/m^2}\ (= 10^{-3}\ \mu\mathrm{bar}).$$

Just as with the above mentioned electrodynamic microphone, this is about 60 dB below 0.10 N/m² or 14 dB above the sound pressure threshold on which the phon-scale is built.

An experimental verification of these considerations is only possible if we can separate the contribution of the amplifier to the noise output. In principle it would be possible to interchange the microphone with a fixed condenser of the same capacity and compare the output signals. We have tried a slightly different way: we left the microphone in its place, but varied its polarizing voltage. The sensitivity of the microphone increases with this voltage. So the noise output of the microphone itself must increase too, and can thus be separated from the constant noise of the amplifier.

The measurements were done in a narrow frequency band (400 c/s width) in the neighbourhood of 9 000 c/s. Figs. 3 and 4 give the results for two

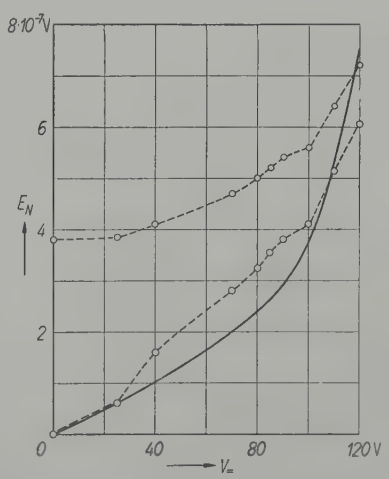


Fig. 3. Experimental and theoretical curves for the noise of one of the two microphones as a function of the bias voltage.

microphones. The upper dotted curve of each figure gives the measured voltage divided by the amplification factor as seen from the microphone. The lower dotted curve has been obtained in the same

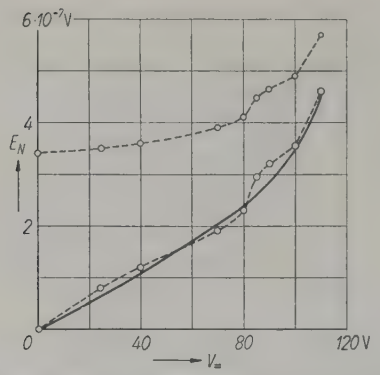


Fig. 4. Experimental and theoretical curves for the noise of the other of the two microphones as a function of the bias voltage.

way, but after correction for the constant amplifier noise, as follows:

$$E^2_{corr}(V) = E^2_{meas}(V) - E^2_{meas}(0).$$

(A separate experiment with a fixed condenser showed that this amplifier noise was indeed independent of $V_=$). Finally, the curves drawn in full in Figs. 3 and 4 are the theoretical ones. To a first approximation they should be straight lines, because we may assume then that the sensitivity of the microphone is proportional to the d.c. voltage, while the noise pressure is independent of it. There is a reason however, why this linear relationship does not hold: the d.c. voltage tends to diminish the distance between diaphragm and electrode. In the first place this means that the sensitivity increases faster than linearly with $V_=$. Secondly the damping coefficient R increases strongly (third power) with decreasing distance, and hence the noise pressure will increase too. Therefore, and in agreement with the experiments, the resulting curves are bent upwards at the higher values of $V_=$.

It seems to us quite certain that these experiments prove the existence of the calculated Brownian motion of the membrane. It is a fascinating idea, that the r.m.s. amplitude over the whole frequency band is about $2 \cdot 10^{-3}$ Å, that is a thousand times smaller than the diameter of an atom, and a hundred times smaller than the amplitude of the temperature vibration of the separate atoms in the diaphragm.

We will finish by showing (Fig. 5) the representation of the electrical impedance of the microphone, as deduced from the electrical and mechanical equations governing its behaviour. This is another way of showing that the microphone must have a self-noise, for there appears a resistive component in its impedance. We have preferred, however, the slightly more fundamental reasoning above, because it leads directly to a noise pressure on the diaphragm, and thus to an easy comparison between theory and experiment.

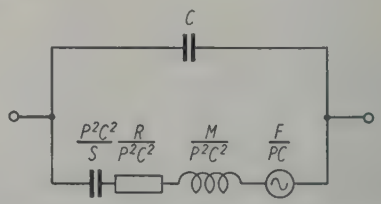


Fig. 5. Equivalent circuit for a condenser microphone.
C = capacity under bias voltage, measured at high frequency;
P = electric field between diaphragm and electrode;
S, R, M = mechanical constants of the diaphragm (see Fig. 2);
F = external force on the diaphragm.

Discussion

MULDERS, C. E.: What is the mechanism of the noise correlated with a mechanical resistance?

LECTURER: In this case a detailed statistical-mechanical analysis of the motion of the air molecules in the narrow gaps behind the diaphragm would certainly give the right fluctuating pressure. But it should not be forgotten that this noise pressure is a fundamental thermodynamic phenomenon which is independent of the material and construction of the resistance, just as in the electrical case.

BERANEK, L. L.: In your paper you mentioned that the noise is flat with frequency. Inspection of your circuit shows that this cannot always be true, because, above resonance, the voltage must decrease at the rate of 12 decibels per octave.

LECTURER: The noise *pressure* has a flat spectrum over the whole frequency range. The noise *voltage* shows a spectrum identical with the pressure characteristic of the microphone, which indeed shows the above mentioned decrease.

LEONARD, R. W.: Can equalization in the following amplifying system retain the good noise factor of an undamped microphone?

LECTURER: Indeed, but of course the equalization of a sharp resonance peak is difficult. This can be avoided (at least in theory) by replacing the mechanical damping by the following system: The output signal of the microphone is amplified, differentiated, and fed back to a polarized grid in front of the membrane. This introduces a damping term in the equation of motion for the membrane, but with little or no noise (so called "cold damping"; see J. M. W. MILATZ a.o., Physica 19 [1953], 195). It should be pointed out that the reduction of noise is not caused by the fact that feed-back is used. (Feed-back can never change a signal-to-noise ratio, as was pointed out in the discussion by S. A. LACEY). The low noise is caused by the removal of mechanical damping. The feed-back serves only to flatten out the frequency characteristic.

CONTRIBUTION À L'ÉTUDE DU FONCTIONNEMENT DE L'OREILLE PAR L'EXAMEN DES COURBES DE RÉPONSE EN ÉCOUTE MONAURALE ET BINAURALE

par R. CHOCHOLLE

Laboratoire de Neurophysiologie du Collège de France, Paris

Sommaire. L'amélioration du seuil en écoute binaurale par rapport au seuil monaural ne peut pas s'expliquer par une hypothèse purement statistique. L'étude des courbes de réponse monaurales ou binaurales montre qu'il ne peut y avoir que sommation au niveau des centres supérieurs. Il y aurait sommation d'effets proportionnels à la puissance, et non à la pression. Si des phénomènes aléatoires conditionnaient le seuil, ceux-ci devraient être assez nombreux: 30 à 40.

Zusammenfassung. Die Herabsetzung der Schwelle beim Hören mit beiden Ohren gegenüber dem Hören mit einem Ohr kann nicht mit einer rein statistischen Hypothese erklärt werden. Die Untersuchung der Erregungskurven beim Hören mit einem und mit beiden Ohren zeigt, daß eine Summierung von Erregungen nur in höheren Nervenzentren stattfindet und dann nur der Schallleistung und nicht dem Schalldruck proportional sein könnte. Wenn aleatorische Phänomene die Schwelle bestimmen, dann müßten sie ziemlich zahlreich sein: 30 bis 40.

Summary. Threshold improvement in binaural listening relatively to monaural listening cannot be explained by a purely statistical hypothesis. A study of monaural and binaural response curves shows that summation only takes place in higher centres. Summation of effects occurs proportional to power, not to pressure. If aleatory phenomena determined the threshold they would have to be numerous: about 30 to 40.

1. Introduction

Des travaux récents ont montré qu'en vision l'amélioration de la sensation en vision binoculaire par rapport à la vision monoculaire était uniquement d'ordre statistique (PIRENNE [3]); en est-il de même en audition? Le seuil est généralement fixé à la valeur d'intensité pour laquelle on obtient 50% de réponses affirmatives par rapport au nombre d'essais effectués. Le calcul montre que la probabilité totale d'entendre devrait être statistiquement augmentée en audition binaurale, ce qui abaisserait le seuil (CHOCHOLLE [1]); si les deux oreilles sont complètement indépendantes l'une de l'autre comme le seraient les oreilles de deux auditeurs différents, la probabilité totale devrait être

$$P = p + p' - pp', \qquad (1)$$

p étant la probabilité d'entendre avec une oreille et p' celle d'entendre avec l'autre oreille.

Si l'amélioration du seuil en écoute binaurale était d'ordre statistique, les courbes cumulatives monaurale et binaurale des fréquences de réponses affirmatives en fonction de l'intensité, qu'on désigne plus simplement sous le nom de *courbes de réponse*, devraient se rejoindre à leurs deux extrémités, comme on le voit d'après la formule (1).

Si, au contraire, les courbes de réponse monaurale et binaurale sont *homologues*, déductibles les unes des autres par translation, la seule conclusion possible est celle d'une sommation au niveau des centres supérieurs: des potentiels d'action seraient encore émis aux intensités légèrement infraliminaires, mais ils ne pourraient franchir les dernières synapses en écoute monaurale; par contre, en écoute binaurale, la sommation qui résulterait de la convergence des voies droite et gauche permettrait le franchissement des dernières synapses.

2. Technique des mesures

La méthode des stimuli constants fut utilisée; le sujet déclenchait lui-même le stimulus; les essais étaient faits alternativement en séries croissantes et décroissantes à des niveaux d'intensité distants de 0,25 dB; en chaque point, 100 essais furent faits, répartis dans 10 séries. On offrait, par ailleurs, de temps à autre un son de même fréquence, mais de 10 dB plus élevé environ. Les fréquences testées furent 200, 1 000 et 10 000 Hz.

Avant chaque série de mesures, l'opérateur déterminait rapidement, par la même méthode que celle utilisée dans les mesures, le seuil de chacune des deux oreilles, ou de l'oreille droite seule s'il s'agissait de mesures monaurales. La valeur liminaire était choisie comme valeur origine, pour l'oreille considérée, de la série en cours de mesures, et notée 0 dB.

Dans le cas de mesures dites avec oreilles *déséquilibrées*, une différence connue d'intensité était introduite entre les deux oreilles, l'oreille droite étant avantagée par rapport à l'oreille gauche; si l'on n'introduisait pas de différences entre les deux oreilles, les oreilles étaient dites *équilibrées*.

Dans les mesures monaurales, j'ai employé une technique analogue à celle utilisée pour les mesures binaurales.

3. Résultats

1) Une série de courbes pour un des sujets à 1 000 Hz est donnée dans la figure 1, à titre d'exemple.

Comme on le voit, les courbes de réponse sont homologues dans toutes les conditions; les courbes de réponse avec oreilles déséquilibrées s'intercalent régulièrement entre les courbes de réponse monaurales et les courbes de réponse binaurales avec oreilles équilibrées. Jusqu'à un déséquilibre de 12 dB, les courbes de réponse avec oreilles déséquilibrées restent significatives.

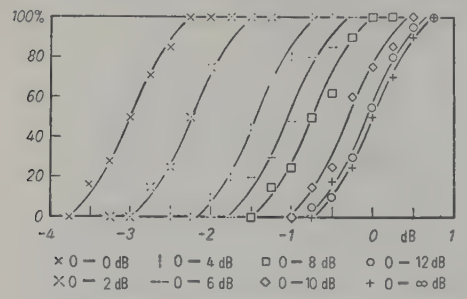


Fig. 1. Pourcentage de réponses affirmatives pour un des sujets en fonction du niveau de l'intensité en dB, en audition monaurale, et en audition binaurale avec oreilles équilibrées et avec oreilles déséquilibrées. Le zéro est la moyenne des seuils dans l'écoute monaurale.

Ainsi, l'amélioration du seuil en écoute binaurale ne peut être due qu'à une *sommation*; elle ne peut être d'ordre statistique.

2) L'amélioration du seuil paraît indépendante de la fréquence, aussi bien avec oreilles équilibrées qu'avec oreilles déséquilibrées, comme on le voit sur la figure 2. Ceci contredit certains travaux antérieurs,

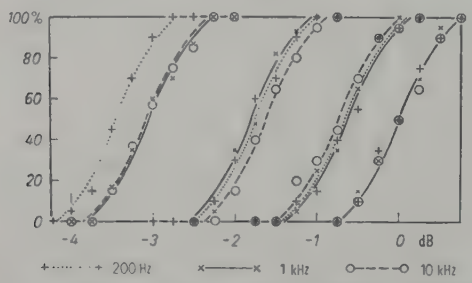


Fig. 2. Comparaison des courbes de réponse à différentes fréquences pour un des sujets, en audition monaurale et en audition binaurale avec oreilles équilibrées et avec oreilles déséquilibrées (0 — 4 dB, 0 — 8 dB).

qui ont été effectués avec d'autres méthodes, il est vrai.

3) L'amélioration du seuil est de 3 dB environ; n'y aurait-il pas sommation d'effets proportionnels à la puissance acoustique? On se rend compte aisément qu'il en est bien ainsi, en comparant les valeurs expérimentales et les valeurs théoriques déduites de cette hypothèse, pour les divers déséquilibres introduits entre les deux oreilles.

Par contre, il n'y a pas sommation d'effets proportionnels à la pression, comme le montrent les calculs, compte tenu du fait que la sommation ne pourrait être que partielle.

Ainsi, le stimulus, aux intensités juxtaliminaires, pourrait bien être une énergie par unité de temps; mais on sait qu'aux intensités élevées, il semble bien que ce soit la pression; il y aurait passage progressif d'un mécanisme à l'autre. Diverses hypothèses peuvent être émises:

a) différences dans la nature du stimulus pour les cellules ciliées des rangées externes et internes de la cochléa; les différences constatées entre ces deux catégories d'éléments sensoriels sont en accord avec une telle hypothèse.

b) évolution différente du potentiel d'action total dans le nerf auditif et de la fréquence des influx dans chaque fibre (Galambos et Davis, [2]).

4) Le stimulus étant une énergie par unité de temps, le stimulus efficace pour atteindre le seuil pourrait être, par exemple, un nombre de chocs de molécules en une unité de temps sur une surface donnée. Aux intensités plus élevées, les chocs de molécules ne seraient plus isolés, il y aurait un véritable bombardement, suivant le rythme imposé par la fréquence du stimulus; ceci serait en accord, en tout cas, avec l'amélioration constatée de 6 dB aux niveaux élevés en écoute binaurale.

Le seuil pourrait encore être lié à d'autres phénomènes quantiques: par exemple, à un nombre minimum d'éléments sensoriels ou de fibres nerveuses en état d'activité. En faisant l'hypothèse que le seuil auditif ne serait atteint que pour un minimum d'événements aléatoires en un temps donné et sur une surface donnée, et que les fluctuations du seuil trouveraient là leur origine, on peut assimiler les courbes de réponse à des courbes représentatives de la loi de Poisson. Le nombre de phénomènes aléatoires devrait être assez grand: 30 à 40, pour les deux meilleurs sujets tout au moins.

5) La pente des courbes de réponse est très raide: il suffit de 1,5 dB pour passer pratiquement de 0% à 100% de réponses affirmatives pour les meilleurs sujets; la pente est pratiquement rectilinéaire de 10% à 90% de réponses affirmatives. Cependant, la représentation par une sigmoïde très redressée paraît plus correcte que la représentation par une droite, d'après les essais de regroupements effectués sur toutes les séries.

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LABORATORY STANDARD MICROPHONES

by R. K. Cook
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Summary. The condenser microphone is successful as a laboratory standard, mainly because it is particularly amenable to calibration by the reciprocity technique. An electro-acoustic analysis of its action is presented here in terms of lumped parameters, which are deduced from measurements of response and electrical admittance. This suggests a new method for extending response determinations to higher frequencies, requiring only admittance measurements and a quarter-wave tube, in addition to a knowledge of the response at lower frequencies.

Sommaire. Le microphone condensateur constitue un standard précieux de laboratoire, surtout parce qu'il est particulièrement susceptible d'être étalonné par la méthode de réciprocité. On analyse ici son fonctionnement électroacoustique à partir des caractéristiques composées qui sont déduites des mesures de la réponse et de l'admittance électrique. Cela suggère une nouvelle méthode pour étendre jusqu'aux fréquences élevées les déterminations de la réponse; pour appliquer cette méthode, il suffit de faire des mesures d'admittance, de disposer d'un tube quart d'onde, et de connaitre la réponse aux basses fréquences.

Zusammenfassung. Das Kondensatormikrophon ist als Eichmikrophon im Laboratorium hauptsächlich deshalb so erfolgreich, weil es sich besonders gut nach der Reziprozitätsmethode eichen läßt. Hier wird eine elektroakustische Analyse seiner Wirkungsweise durchgeführt, bei der man sich zusammengesetzter Parameter bedient, die aus Messungen der Frequenzcharakteristik und der Admittanz hergeleitet werden. Dies gab die Anregung zu einer neuen Methode, die Frequenzkurven im Bereich höherer Frequenzen aufzunehmen, wozu man nur Admittanzmessungen und ein Viertelwellenlängen-Rohr benötigt, wenn die Frequenzkurve bei tiefen Frequenzen bekannt ist.

Research and engineering laboratories working in various phases of acoustics require standard microphones for measuring sound in air, water, and other fluids over wide frequency ranges. We confine our attention here to microphones suitable for measurement of sound in air over the frequency range from about 30 to 20 000 c/s.

One sometimes thinks of sound as being caused by air molecules in vibratory movement. However, no satisfactory microphone for measurement of air molecular motions has ever been devised, although the molecular displacements can be measured with illuminated smoke particles, and the molecular velocities can be measured with a RAYLEIGH disk.

Nearly all the microphones which have ever been devised respond to the sound pressure associated with the molecular motions, as does the ear of man. Absolute measurements of sound pressure are of basic importance for research in acoustics. They are most conveniently made by means of standard microphones of special design which lend themselves to calibration by the reciprocity technique. Other practical requirements on standard microphones are freedom from non-linear distortion, uniformity of response of both amplitude and phase as a function of frequency, stability of response (including freedom from hysteresis) with temperature and humidity changes, and physical size small enough to minimize the amount of scattered sound power.

The condenser microphone appears to meet these requirements more satisfactorily than any other type of microphone. A knowledge of its construction is not sufficient to yield with requisite accuracy an absolute value for its response, which is best determined by means of the reciprocity method. Nevertheless, we can determine empirically simple lumped parameters which describe very well the microphone's response and electrical admittance as a function of frequency, as we shall see presently.

We consider the electro-acoustic action of a condenser microphone (see Fig. 1) as follows. The

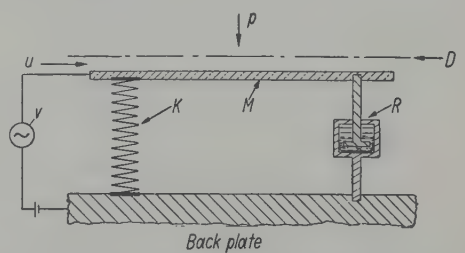


Fig. 1. Schematic diagram of condenser microphone action. p = sound pressure; u = volume displacement of diaphragm; D = diaphragm equilibrium; v = a.c. voltage; K = spring constant; M = equivalent mass of diaphragm; R = viscous resistance; C_0 = capacitance with diaphragm at equilibrium; Φ = electromechanical coupling constant.

simplest assumption is that the action can be described by the generalized coordinates u (the volume displacement of the microphone diaphragm from its zero-signal position) and q (the difference between the electrical charge on the condenser and its zero-signal charge). The potential energy W, kinetic energy T, and dissipation function D are

$W = q^2/2C_0 - \Phi qu + Ku^2/2 +$ (terms of higher order)

$$T = M\dot{u}^2/2, \quad D = R\dot{u}^2/2 \tag{1}$$

in terms of the generalized coordinates u and q. These quantities are as defined in Fig. 1.

The Lagrangian equations of motion for the system are obtained from

$$\frac{\mathrm{d}}{\mathrm{d}t}\left(\frac{\partial T}{\partial \dot{u}}\right) - \frac{\partial T}{\partial u} + \frac{\partial W}{\partial u} + \frac{\partial D}{\partial \dot{u}} = p$$

and from a similar equation for the coordinate q. Therefore the generalized forces p (the sound pressure on the diaphragm) and v (the variable part of the potential difference between diaphragm and backplate) are given by

$$\begin{aligned} M\ddot{u} + R\dot{u} + Ku - \Phi q &= p, \\ -\Phi u + q/C_0 &= v. \end{aligned} \tag{2}$$

If the generalized forces and coordinates vary like $e^{j\omega t}$, then

$$\begin{aligned} (-\omega^2 M + j\omega R + K)\, u - \Phi q &= p, \\ -\Phi u + q/C_0 &= v. \end{aligned} \tag{3}$$

If $p = 0$, the admittance of the microphone as seen from its electrical terminals is by (3)

$$Y = j\omega C_0 + \\ + j\omega\Phi^2 C_0^2/(-\omega^2 M + j\omega R + K - \Phi^2 C_0). \tag{4}$$

The response ϱ is the electromotive force (or open-circuit voltage) per unit sound pressure applied to the diaphragm. From eq. (3) we find that the voltage v at the microphone's terminals, when the electric current (and hence q) is zero, is

$$v = -\Phi p/(-\omega^2 M + j\omega R + K).$$

This voltage has the opposite sign of the electromotive force, and therefore the response is

$$\varrho = \Phi/(-\omega^2 M + j\omega R + K). \tag{5}$$

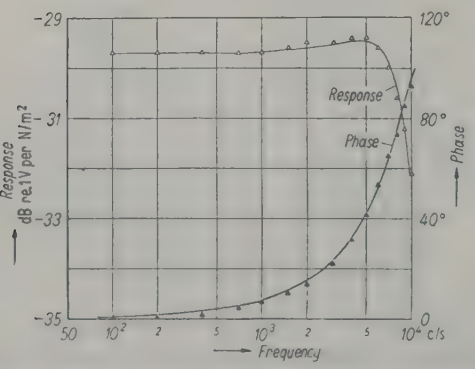


Fig. 2. Response of type 640AA condenser microphone No. 904. △ = amplitude, and ▲ = phase of response measured at the Electrical Communication Laboratory, Tokyo, Japan. Curves are computed from eq. (5) with the following lumped parameters:
$K = 1.52\times10^{12}$ N/m^5, $R = 3.16\times10^7$ N s/m^5,
$M = 434$ kg/m^4, $\Phi = 4.95\times10^{10}$ V/m^3.

Fig. 2 gives a comparison between the response measured by the reciprocity method, and the response computed from the lumped parameters tabulated in the figure caption. Fig. 3 is a similar comparison for the same microphone between the measured electrical admittance (found with a SCHERING bridge) and the computed admittance. The lumped parameters are found as follows: Φ/K is the same as the response at low frequencies; Φ/R is

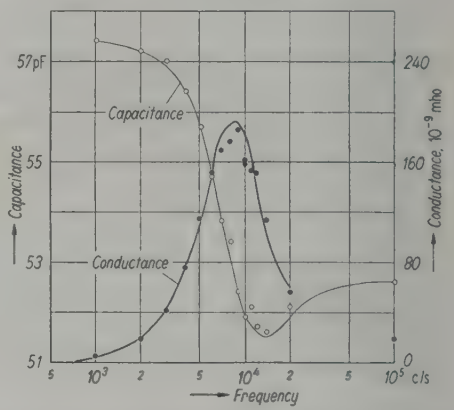


Fig. 3. Electrical admittance of microphone No. 904. ○ = capacitance, and ● = conductance, both measured with a SCHERING bridge. Curves are computed from eq. (4) with the same lumped parameters as in Fig. 2, and also $C_0 = 52.6\times10^{-12}$ F.

the maximum measured value of $|\omega\varrho|$, which occurs at a measured pulsatance ω_0 such that $K = \omega_0^2 M$; C_0 is the capacitance of the microphone with the diaphragm blocked, and therefore is the measured value either at high frequencies, or near resonance; the capacitance measured at low frequencies is $C_0/(1-\Phi\varrho C_0) = C_0 K/(K-\Phi^2 C_0)$. These five relationships yield the five lumped parameters, K, M, R, Φ and C_0. The good agreement shown in Fig. 2 and 3 has been obtained for other microphones of the same type.

Equations (4) and (5) can be rewritten so that the mechanical quantity $-\omega^2 M + j\omega R + K$ is replaced by Z, and Z may vary with ω in any arbitrary fashion. Thus

$$\begin{aligned} Y &= j\omega C_0 + j\omega\Phi^2 C_0^2/(Z - \Phi^2 C_0), \\ \varrho &= \Phi/Z. \end{aligned} \tag{6}$$

This suggests a new method for calibration of condenser microphones, which has been investigated by Messrs. SIMMONS and BIAGI of the staff of the National Bureau of Standards. Precision measurements, which can be made with a SCHERING bridge, of the admittance Y and the blocked admittance $j\omega C_0$ will yield Φ^2/Z at any frequency. Apart from a factor Φ, this is the same as ϱ. The expression for ϱ is

$$\varrho = (Y - j\omega C_0)/\Phi C_0 Y. \tag{7}$$

In practice, both Φ and C_0 are obtained from response and admittance measurements made at frequencies below 10 000 c/s. Eq. (7) then serves to extrapolate ϱ to higher frequencies.

Fig. 4 shows a comparison between the response determined with this new method and the response

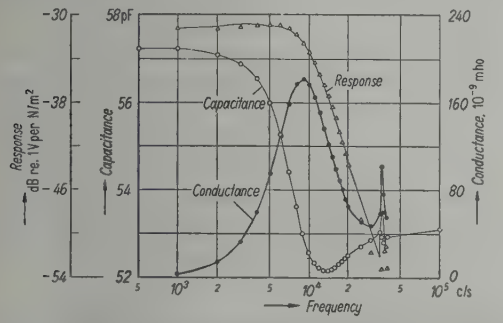


Fig. 4. Admittance and response of type 640 AA condenser microphone No. 1087. ○ and curve = capacitance, and ● and curve = conductance, both measured with a SCHERING bridge. △ = response measured by the reciprocity method. Curve for response is computed from admittance measurements by means of eq. (7), and is fitted to the measured response at frequencies below 10 000 c/s by adjustment of ΦC_0.

determined by the usual reciprocity method. The latter required a hydrogen-filled 3.18 cm³ coupler at frequencies above 10 000 c/s. The only acoustical equipment required during the electrical admittance measurements, however, was an air-filled quarter-wave tube (for elimination of air loading on the diaphragm). The good agreement between the two response determinations shown in Fig. 4 indicates that absolute pressure responses of condenser microphones can be determined to within about 0.2 dB for frequencies as high as 20 000 c/s.

The author is indebted to Mr. F. BIAGI for help in the experimental measurements and calculations.

Discussion

NIELSEN, A. K.: If a very high level is employed at the source microphone the system is non-linear. But it appears that the reciprocity calibration gives correct results if the distortion products are filtered out.

LECTURER: This is correct for the distortion products which appear when the amplitude of the alternating driving voltage approaches the polarizing voltage. But there might be other sources of distortion. For example, the displacement of a plate-diaphragm is a non-linear function of the force which deflects it. Experimentally, this seems to affect the response by not more than approximately 0.1 dB.

RAY, S. N.: Would Dr. COOK kindly comment on the suitability of the hot-wire type of microphone for measurement of particle velocity?

LECTURER: If a hot-wire microphone is to be used for quantitative measurement of particle velocity, the experimenter must employ a carefully controlled polarizing current of air, because the response (voltage output per unit particle velocity) is roughly proportional to the velocity of the polarizing current. As far as I know, such control has never been satisfactorily accomplished. Also, the response is affected by the adsorption by the hot-wire of constituents of the atmosphere. The hot-wire microphone is frequently used as an indicator, but does not appear to be suitable for quantitative measurement of air particle velocity.

RADEMAKERS, A.: Concerning a remark of Dr. P. BRÜEL: A condenser microphone in a normal L.F.-amplifier works with constant charge. As $C = 1/d$ and $V = Q/C$ there is no distortion at all if there is no stray capacity. For the same reason there is no influence of the bias voltage on the stiffness: the constant charge gives a constant pulling force, independent of the distance.

RECENT WORK AT THE NATIONAL PHYSICAL LABORATORY ON THE FREE-FIELD CALIBRATION OF STANDARD MICROPHONES

by R. S. DADSON and E. G. BUTCHER
National Physical Laboratory, Teddington, England

Summary. Recent developments at the National Physical Laboratory on free-field standards of sound pressure and microphone calibration are described. In the low-frequency region free-field plane-wave conditions are difficult to achieve in conventional absorbent rooms, and this has been met by the installation of a large duct in which truly plane progressive waves may be generated down to 25 c/s. At the higher frequencies the RAYLEIGH disc and reciprocity free-field standards have been compared and show excellent agreement.

Sommaire. On décrit les récents travaux effectués au National Physical Laboratory sur la réalisation des standards de pression sonore à l'air libre et l'étalonnage des microphones. En basse fréquence, il est difficile de satisfaire aux conditions d'onde plane en champ libre dans les chambres absorbantes; on y est parvenu en installant un large conduit dans lequel on peut produire, jusqu'à une fréquence minima de 25 Hz, des ondes progressives réellement planes. On a comparé aux fréquences élevées le standard obtenu avec le disque de RAYLEIGH et celui en champ libre suivant la méthode de réciprocité; l'accord est excellent.

Zusammenfassung. Es werden die letzten Entwicklungen auf dem Gebiet der Messung absoluter Werte des Schalles im freien Schallfeld und der Mikrophoneichung im National Physical Laboratory beschrieben. Da man bei tiefen Frequenzen in den üblichen schallabsorbierenden Räumen

nur schwer ebene Wellen wie im freien Schallfeld herstellen kann, wurde ein großes Rohr aufgebaut, in dem wirklich ebene fortschreitende Wellen bis hinunter zu 25 Hz erzeugt werden können. Bei höheren Frequenzen werden Meßmikrophone, die mit der RAYLEIGHscheibe und nach der Reziprozitätsmethode geeicht waren, miteinander verglichen. Es ergab sich eine ausgezeichnete Übereinstimmung.

1. Introduction

In the course of a "Discussion on Units and Standards" arranged by the National Physical Laboratory in 1946, an account was given by one of the present authors of the position of acoustical standards of measurement at the N.P.L. at that time [1]. Much of that account is still applicable, but a number of improvements have been made in the meantime which are relevant to the objects of this Congress, and this paper gives a brief description of two developments to which considerable attention has been given. Before considering these in further detail it may be of value to list the basic standards currently used at the N.P.L. for different ranges of frequency. At low frequencies free-field calibrations are obtained directly by comparison with the RAYLEIGH disc in a plane progressive wave system in a duct, or alternatively (for standard pressure-type microphones) are derived from pressure calibrations based on the RAYLEIGH disc, the pistonphone, or the method of reciprocity. At the higher frequencies use is made either of the RAYLEIGH disc, or reciprocity, in a progressive wave system. It has consistently been the aim at N.P.L. to have two independent methods giving results in satisfactory agreement over each part of the frequency range, so that if trouble is suspected an entirely independent check is always available.

2. Free-field calibration at low frequency

Recent work at the N.P.L. over the low-frequency range has largely been concerned with the problem of generating an accurately plane progressive wave system for the free-field calibration of standard microphones, and substandard microphones of all kinds including pressure-gradient and other directional types. The conventional absorbent room has evident limitations as regards the very low frequencies. In the first place the available absorbents, though greatly improved recently, are not adequate to give truly reflectionless conditions at the lowest frequencies. Secondly as regards the simulation of plane progressive waves, it is not possible in rooms of normal dimensions to conduct measurements at a sufficient distance from the source to obtain the correct relations between sound pressure and particle velocity. Thirdly, spherical radiation imposes limitations on the magnitude of sound pressure which is obtainable at low frequencies with normal loudspeakers. The second limitation is of especial importance in dealing with the calibration of pressure gradient microphones, and more especially of types relying on a combination of pressure and pressure-gradient elements. These difficulties can, however, all be met by the use of a solidly constructed duct with a good absorbent termination. Fig. 1 shows the essential features of an apparatus of this kind which has recently been installed at the National Physical Laboratory. The absorbent section consists of a wedge-shaped structure of glass fibre blankets some 7 m in length which has been shown to reduce the reflected sound amplitude to about 1% over the useful range of frequency, 25 to 450 c/s. Accurate plane wave conditions are established within about a metre of the loudspeaker and are maintained over the working section of the duct. The apparatus is convenient not only for the establishment of accurately plane progressive waves for standardisation purposes, but also for many sub-standard measurements, such as the determination of the sensitivity and directional properties of pressure-gradient, or composite microphones in which an accurately defined wave system is essential. It is also easily adapted to the determination of the normal impedance of comparatively large samples of absorbing materials or constructions. A full description of this equipment will shortly be published elsewhere.

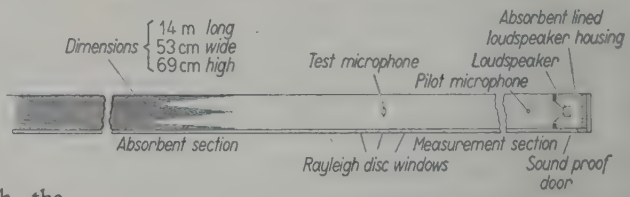


Fig. 1. Plan of duct for acoustical measurements at low frequencies.

3. Free-field calibration at high frequency

Over the higher frequency range both the RAYLEIGH disc and reciprocity standards are in use for the free-field calibration of the Laboratory's standard microphones, and the accurate comparison of these standards is thus of considerable importance. The results shown in Fig. 2 illustrate the calibration of one of the small condenser microphones which are maintained as the Laboratory's standards for free-field work. The RAYLEIGH disc calibration has been carried out from 30 to 25 000 c/s, and the

direct reciprocity calibration from 1000 to 100 000 c/s, the range of overlap of the two methods being thus from 1000 to 25 000 c/s. Over this range the agreement between the two methods at individual frequencies is of the order $\pm 2\,\%$, and the average discrepancy less than 1 %. The RAYLEIGH discs used at the Laboratory are normally 1 cm in diameter

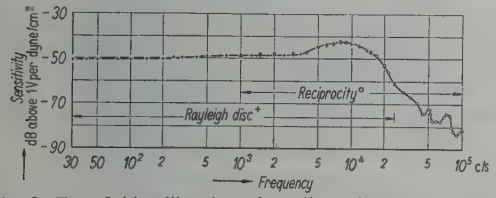


Fig. 2. Free field calibration of small condenser microphone up to ultrasonic frequencies. Comparison of RAYLEIGH disc and reciprocity calibrations.

for frequencies up to about 3 000 c/s and 0.5 cm for the higher range.

In free-field work such consistency can only be obtained if great care is taken over all the experimental conditions, and in work of this order of accuracy there is probably little to choose, either in accuracy or convenience, between the two methods. The general conclusion is therefore that the Laboratory's standards of free-field calibration are established to an accuracy of about 1 % over the frequency range covered.

The work described in this paper was carried out as part of the research programme of the National Physical Laboratory and the paper is published by permission of the Director of the Laboratory.

Reference

[1] DADSON, R. S., Discussion on Units and Standards, Proc. roy. Soc. A. 186 [1946], 149.

Discussion

LEEUWEN, F. J. VAN: How is the distance measured in your reciprocity measurement?

LECTURER: The separation distance in the reciprocity free-field measurements is strictly the radius of the spherical wave; at the distances (30 ··· 50 cm) adopted in the measurement this is equal to the distance between the microphone diaphragms to well within the accuracy required.

GAVREAU, V.: Dans certaines conditions l'absorption des ultra-sons dans l'air est très faible: nous avons pu détecter des sons de 75 kHz à plus de 14 m et des sons de 150 kHz restaient très intenses à 3 m. Ne pourrait-on pas éliminer les erreurs dues à l'absorption en desséchant l'air et en supprimant le CO_2 dans la chambre utilisée pour ces mesures?

LECTURER: Lors de nos mesures l'absorption du son par l'air était en général négligeable car les distances entre le microphone et le haut-parleur n'étaient pas très grandes; même aux fréquences les plus élevées (100 kHz), la correction était toujours petit et pouvait se déterminer assez précisément au moyen de mesures à une série de distances.

ÉTALONNAGE ABSOLU DE MICROPHONES AUX FREQUENCES AUDIBLES ET ULTRA-SONORES

par V. GAVREAU

Centre de Recherches Scientifiques, Industrielles et Maritimes, Marseille

Sommaire. Le pistonphone ultra-sonore est constitué par la base supérieure d'un cylindre métallique massif vibrant en demi-longueur d'onde. L'intensité du son est calculée à partir de l'amplitude de vibration mesurée optiquement ou à partir de la pression de radiation déterminée par une pesée. La méthode de réciprocité utilisée aux fréquences audibles élevées, nécessite l'emploi d'une chambre de compression fermée par un capuchon à petit trou. La formule classique est modifiée pour tenir compte de l'effet de baffle.

Zusammenfassung. Als Ultraschallpistonphon wird die Stirnfläche eines massiven Metallzylinders benutzt, der eine halbe Wellenlänge lang ist. Die Schallintensität wird aus der optisch bestimmten Schwingungsamplitude und aus dem mit der Waage gemessenen Strahlungsdruck berechnet. Für die im Bereich der höheren Hörfrequenzen benutzte Reziprozitätsmethode benötigte man eine Druckkammer, deren Verschluß aus einer mit einem Loch versehenen Kappe bestand, die bei höheren Frequenzen als „Schallwand" wirkt. Um diesen Effekt zu berücksichtigen, wird die klassische Formel abgeändert.

Summary. The ultrasonic pistonphone is formed of the upper end of a massive metallic cylinder vibrating at a half wavelength. The intensity of the sound produced is calculated either from the amplitude of vibration optically measured or from the pressure of radiation measured on a balance. The use of the reciprocity method at high audio-frequencies involves the use of a pressure chamber closed by a cap with a small hole. The classical formula is modified to take account of the effect of the baffle.

1. Introduction

Un microphone étalonné doit permettre de mesurer le champ sonore qu'on aurait en son absence au point où on effectue cette mesure. Or, la caractéristique de sensibilité ainsi définie, d'un microphone, diffère considérablement de celle qu'on obtient par un étalonnage dit "à pression constante" car aux fréquences élevées elle dépend de la forme et des dimensions du microphone.

Au Laboratoire d'Électro-Acoustique du Centre de Recherches Scientifiques, Industrielles et Maritimes de Marseille, nous avons mis au point des méthodes permettant l'étalonnage absolu de microphones dans les conditions normales de leur emploi ("à champ libre") pour des fréquences pouvant atteindre 100 kHz.

2. Pistonphone ultra-sonore

Jusqu'à 10 kHz on peut utiliser un piston épais en duralumin collé sur une couche de caoutchouc. Pour des fréquences plus élevées, nous utilisons un cylindre vibrant longitudinalement en demi-longueur d'onde [1] dont la base supérieure vibre comme un piston (Fig. 1). Cette base, soigneusement polie, constitue

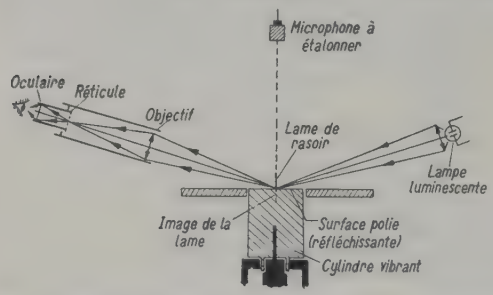


Fig. 1. Schéma de principe de l'étalonnage d'un microphone en utilisant un pistonphone ultra-sonore.

un miroir dans lequel se forme l'image d'un tranchant de lame de rasoir fixé à faible distance au-dessus du piston vibrant: l'image de la lame vibre avec une amplitude double de celle du piston. On mesure cette amplitude à l'aide d'un dispositif optique spécial comportant un objectif de 80 mm de diamètre et 400 mm de distance focale et un oculaire de microscope à fort grossissement muni d'un réticule micrométrique. Cet appareil permet la mesure de déplacements de quelques centièmes de millimètre à une distance de 40 cm, donc sans perturber le champ ultra-sonore. Un éclairage stroboscopique par un Glow Modulator Tube (Sylvania), commandé par une hétérodyne auxiliaire, facilite les mesures.

On calcule l'intensité du champ sonore en appliquant la formule connue [2]

$$I = 2\,\varrho c\,\dot{\xi}^2 \sin^2 \frac{\pi m^2 \lambda}{2r} \qquad (1)$$

où ϱ = la densité de l'air; c = la vitesse de propagation du son; $\dot{\xi} = 2\pi f \xi$ = la vitesse du piston; ξ = l'amplitude du piston; f = la fréquence; $m = a/\lambda$; a = le rayon du piston; λ = la longueur d'onde du son; r = la distance du microphone au piston. On déduit de l'éq. (1) la pression acoustique

$$p = \sqrt{\varrho c\, I}.$$

Un autre procédé très commode et utilisable même aux fréquences très élevées, consiste à mesurer la pression de radiation sonore et d'en déduire l'intensité acoustique par la formule de Lord Rayleigh [3]

$$p_1 - p_0 = E\left(\frac{1}{3} + \frac{\gamma - 1}{2}\right)$$

où $p_1 - p_0$ est la pression de radiation, E la densité moyenne d'énergie acoustique totale et $\gamma = 1{,}40$ pour l'air.

En y substituant

$$p_1 - p_0 = Mg/S \cos^2 \theta,$$

Mg étant le poids nécessaire pour équilibrer la pression de radiation contre un cône de base S dont la génératrice fait un angle θ avec la base, et

$$I = E\,c,$$

on obtient

$$I = M \frac{g\,c}{S \cos^2 \theta\,[1/3 + (\gamma - 1)/2]} \qquad (2)$$

d'où $$p = \sqrt{\varrho c\, I}.$$

Pour un microphone électrodynamique L.M.T. nous avons trouvé par le procédé optique une sensibilité $s = 1{,}69$ mV/Nm^{-2} (moyenne des mesures) et par le procédé de la pression de radiation $s = 1{,}71$ mV/Nm^{-2} (moyenne des mesures) à 9,45 kHz.

Cette concordance des résultats constitue une vérification expérimentale de la validité de la formule théorique de Lord Rayeigh (éq. (2)), ce qui permet l'emploi de pistonphones à cylindre vibrant pour l'étalonnage absolu de microphones jusqu'à 100 kHz. Nous avons réalisé des cylindres vibrants de 9,45 kHz, 38,8 kHz, 75 kHz et 90 kHz.

La mesure optique devient difficile pour les fréquences supérieures à 40 kHz, même lorsqu'on remplace l'oculaire par un petit microscope auxiliaire.

3. Méthode de réciprocité

Les mesures étaient faites dans une grande chambre sourde en utilisant comme transducteur auxiliaire une chambre de compression de haut-parleur fermée par un capuchon avec un trou de 8 mm de diamètre (Fig. 2).

a) En utilisant comme source le haut-parleur de la grande chambre sourde, on trouve les tensions V_1 et V_2 aux bornes du transducteur et du microphone;

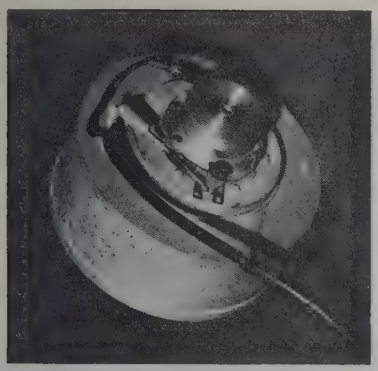


Fig. 2. Chambre de compression à ouverture réduite utilisée comme transducteur auxiliaire.

b) en utilisant le transducteur comme émetteur de son, $I_1 = U_1/R$ étant le courant dans sa bobine mobile, mesuré dans une résistance $R = 20$ ohm connectée en série avec elle, on trouve une tension U_2 aux bornes du microphone placé à une distance $r = 110$ cm du transducteur. La sensibilité du microphone est donnée par la formule connue [4]

$$s_2 = \sqrt{\frac{V_2}{V_1} \cdot \frac{U_2}{I_1} \cdot \frac{2r\lambda}{\varrho c}}. \qquad (3)$$

Cette formule a été établie en assimilant le transducteur à une petite sphère pulsante de rayon a et d'impédance acoustique

$$z_r = \varrho c/2a\lambda. \qquad (4)$$

Ce qui n'est plus vrai pour les fréquences élevées, lorsque le boitier du transducteur joue le rôle d'un "baffle": le transducteur pourrait alors être assimilé à un piston dans un mur plan dont l'impédance acoustique est:

$$z_r = \frac{z_s}{S} = \frac{z_s}{\pi a^2} = \frac{\varrho c}{a^2} \sqrt{\left[1 - \frac{J_1(2ka)}{ka}\right]^2 + \left[\frac{K_1(2ka)}{2k^2a^2}\right]^2}.$$

Pour pouvoir plus facilement comparer les résultats avec ceux obtenus dans le cas de la sphère pulsante, nous écrirons cette formule sous la forme:

$$z_r = \left(\frac{\varrho c}{a\lambda}\right) \frac{\lambda}{\pi a} \sqrt{\left[1 - \frac{J_1(2ka)}{ka}\right]^2 + \left[\frac{K_1(2ka)}{2k^2a^2}\right]^2} \qquad (5)$$

où $k = 2\pi/\lambda$. J_1 et K_1 sont des fonctions de BESSEL et a est le rayon du trou, 0,4 cm dans notre cas. Pour la fréquence la plus élevée utilisée au cours de ces mesures $f = 15\,000$ Hz, la formule (5) donne:

$$z_r = \frac{\varrho c}{a\lambda} \cdot \frac{1}{0{,}964}. \qquad (6)$$

En comparant cette expression à l'expression (4), on voit que dans notre cas particulier, z_r tend vers une valeur deux fois plus grande dans le cas des fréquences élevées. Donc la sensibilité devrait être divisée par $\sqrt{2}$ pour les fréquences élevées [5].

Nous avons corrigé les valeurs trouvées pour la sensibilité en utilisant la formule:

$$s_{corrigée} = y s_2$$

où y est un coefficient numérique dont la valeur (déterminée pour plus de commodité graphiquement) décroit 1 à 1 Hz jusqu'à $1/\sqrt{2} = 0{,}707$ à 15 kHz.

Le microphone étalonné par ce procédé est utilisé pour des mesures des seuils d'audibilité, et les valeurs trouvées concordent bien avec les valeurs trouvées aux États-Unis dans les mêmes conditions.

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[2] DAVIS, A. H., L'acoustique moderne. Éd. Dunod, Paris 1936, p. 76 *). Formules plus précises, tenant compte des dimensions finies du "baffle" de TADAMOTO NIMURA et YOSHIYUKI WATANABE dans J. acoust. Soc. Amer. 25 [1953], 76.

[3] RAYLEIGH, Lord, Phil. Mag. [1905] S. 6, 10 (2), pp. 364 à 374. Nous avons, bien entendu, corrigé la faute d'impression en remplacant le signe — par le signe + dans l'éq. (41) comme on le voit immédiatement d'après les équations (39) et (40) dont est déduite l'éq. (41).

[4] BERANEK, L. L., Acoustic measurements. J. Wiley & Sons, New York 1949, pp. 113 à 121.

[5] On trouve les expressions de l'impédance d'un disque avec un baffle circulaire de dimensions finies dans l'article de T. NIMURA et Y. WATANABE [2]: on pourrait les utiliser pour un calcul plus complet.

Discussion

DADSON, R. S.: La formule actuellement admise pour la pression de radiation sonore, n'est pas la formule plus compliquée de Lord RAYLEIGH, considérée actuellement comme incorrecte (voir C. SCHAEFER, Ann. Phys., Lpz. 35 [1939], 473), mais la formule plus simple, établie antérieurement par LARMOR et Lord RAYLEIGH.

CONFÉRENCIER: Il existe en effet, plusieurs formules différentes donnant la pression de radiation, et je ne sais pas laquelle de ces formules est correcte. Je serais très reconnaissant pour toutes les références bibliographiques relatives à leur discussion. Toutefois, nos essais paraissent confirmer la formule établie par Lord RAYLEIGH en 1905.

NIELSEN, A. K.: La formule correcte donnant la pression de radiation sonore est celle de LARMOR: $p_r = 2E$ et non pas la formule établie par Lord RAYLEIGH en 1905. Ceci a été confirmé expérimentalement au Danemark par le Prof. HARTMANN. La question a fait l'objet d'une discussion très détaillée publiée dans Annalen der Physik.

CONFÉRENCIER: Nos essais paraissent, au contraire, confirmer la formule de Lord RAYLEIGH. Ceci s'explique peut-être par le fait que dans le cas de sons très intenses, la loi de compression non-adiabatique ne peut pas être négligée.

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*) La formule indiquée dans cette communication correspond au cas d'un microphone assez éloigné du pistonphone. Comme l'étalonnage du microphone et les mesures de la pression de radiation sonore ont été faits à une petite distance du piston vibrant, une formule plus correcte serait [2]:

$$I = \tfrac{1}{2}\varrho c \dot{\xi}^2 \left[1 - \frac{J_1(ka)}{ka}\right]$$

qui se réduit à: $I = \frac{1}{2}\varrho c \dot{\xi}^2$ dans le voisinage immédiat du piston vibrant.

Ceci expliquerait les valeurs trop petites que nous avions trouvées pour la pression de radiation sonore, valeurs qui semblaient confirmer la troisième formule de Lord RAYLEIGH.

A METHOD FOR MEASURING SOUND ABSORPTION IN THE FIELD

by C. W. KOSTEN

Laboratory for Technical Physics, Technological University, Delft

Summary. The arrangement is essentially a normal standing wave tube held against the wall to be tested. Leakage of sound through the absorbing layer is prevented by surrounding the wave tube by another wave tube of greater diameter, properly actuated in the same frequency. The criterion for "no leakage" is equality of sound pressure in both tubes near the sample. Preliminary results given corroborate the idea that the method is reliable.

Sommaire. Le dispositif employé se compose essentiellement d'un tube normal à onde stationnaire qui est appliqué contre la paroi à essayer. On évite les fuites acoustiques dans le revêtement absorbant en entourant ce tube d'un second tube de diamètre plus grand, convenablement excité à la même fréquence. Il n'y a pas de fuite si la pression sonore est la même dans les deux tubes, au voisinage de l'éprouvette. Les résultats préliminaires indiqués montrent que cette méthode est sûre.

Zusammenfassung. Die Meßanordnung besteht im wesentlichen aus einem normalen Kundtschen Rohr mit stehenden Wellen, das gegen die zu messende Wand gehalten wird. Um eine Undichtigkeit bei Messung absorbierender Schichten zu vermeiden, wird das Meßrohr von einem zweiten mit größerem Durchmesser umgeben, das in passender Weise mit der gleichen Frequenz betrieben wird. Als Kriterium dafür, daß kein Leck vorhanden ist, benutzt man die Gleichheit des Schalldrucks in beiden Rohren nahe der Probe. Die hier angegebenen vorläufigen Resultate scheinen die Zuverlässigkeit der Methode zu bestätigen.

Numerous well-known methods exist for measuring the sound absorption coefficient in the laboratory. Although field measurements are sometimes badly needed there does not seem to exist a reliable method for this purpose. It should be very important that the consulting engineer or the architect should be able e.g. to verify that a certain absorbing wall or ceiling meets the requirements agreed upon or to measure on the spot the possibly detrimental effect of a certain system of decoration. Such questions are of considerable economic interest and justify efforts to develop a method of measurement in the field, i.e. a method that enables the operator to measure the absorption coefficient of a surface without taking a sample or damaging the surface.

Two field methods have been described in the literature, one by LOYE and MORGAN [1] (Fig. 1), the other by JONES, EDELMAN and LONDON [2] (Fig. 2). The apparatus used by JONES et al. is essentially a normal interferometer or standing-wave tube bonded to the layer to be tested. If there should be no leakage of sound the apparatus would yield the absorption coefficient at normal incidence. The apparatus of LOYE and MORGAN may be described as an interferometer of negligible length. A sound source F actuates a diaphragm just in front of the layer being tested, while a pressure microphone M measures the pressure level in the cavity. The level difference, as compared to the level found with a completely reflecting wall, is used as a measure of the absorption. This method neglects the complex character of the wall impedance and suffers, as does JONES' method, from possible leakage of sound.

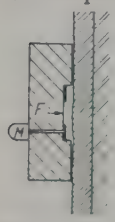


Fig. 1. LOYE and MORGAN's set-up for measuring in the field [1].

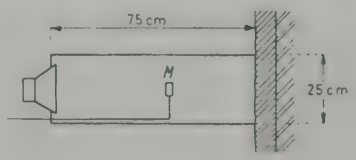


Fig. 2. JONES, EDELMAN and LONDON's long-tube method for field measurements [2].

From a theoretical point of view it is very questionable whether leakage of sound can be sufficiently avoided with the methods as described. In an earlier publication a series of absorption characteristics was given for absorbing layers of different thickness, air resistance and structure factor [3]. From these curves one sees at a glance that layers with a specific air resistance of 10^6 or 10^7 MKS-rayl $\times$ meter are poor absorbers at audio frequencies. The sound waves only penetrate a few millimeters into the material, clearly demonstrated by the fact that a layer of 1 cm absorbs almost as much as a layer of infinite thickness, whilst resonance frequencies in layers of 1 cm and more are hardly to be noticed. Layers of such great airtightness are not subject to much lateral spreading of sound when measured with one of the methods described, but they are of little interest. The good absorbers behave just the other way round. They have specific air resistances of only 10^4 to 10^5 rayl $\times$ meter, show up resonances when applied in thicknesses of up to even 10 cm, are therefore accessible to their full depth by the sound waves, which is also clear from the fact that the absorption characteristic changes with the thickness even up to very great

thicknesses. Therefore good absorbers are so easily permeable to sound that lateral spreading through them must take place to a considerable degree.

We therefore came to the conclusion that field measurements of this type can only be carried out with sufficient accuracy if leakage of sound is duly prevented. To this end a double interferometer has been constructed (see Fig. 3) consisting of two co-

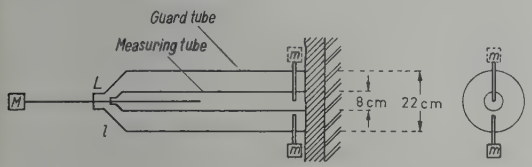


Fig. 3. The new method with a guard tube.

axial cylinders of which the outer cylinder acts as guard cylinder to prevent leakage out of the inner cylinder, the measuring tube. It is clear that in this way leakage of sound can be avoided exactly and that equality of sound pressure just inside and outside the measuring tube just in front of the layer under test is a good criterion for compensation. The two sound pressures ought to be equal of course both in magnitude and in phase. We therefore need two sound sources of the same frequency, one of which is adjustable in magnitude and phase. Moreover a differential microphone in the wall of the inner cylinder is necessary for indicating the sound pressure difference. The preliminary set-up available was simply operated with a probe microphone alternatively connected to the inner and outer cylinder.

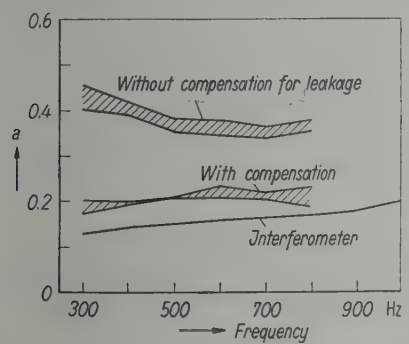


Fig. 4. Experimental results with the new method on an inflexible layer of 2.1 cm thickness. For comparison the results obtained in a normal interferometer.

In Figs. 4 and 5 a few results are given *). Fig. 4 relates to a layer of Faburtex, flax straw material, 2.1 cm thick, specific air resistance $4 \cdot 10^5$ rayl $\times$ meter. The curves "without compensation" and "with compensation" have been measured with the apparatus of Fig. 3 without and with the use of the guard wave field in the outer cylinder. In order to have a check as to the accuray of the "compensated" measurement a sample was sawn out of the plate tested, the sample then being measured along conventional lines in a normal interferometer. The mounting conditions were chosen as closely the same as possible.

Whereas the material to which Fig. 4 relates is nearly inflexible, the material of Fig. 5 is very soft indeed (Kramfors, bonded woodfibres, 60 kg/m³, 3.5 cm thick, specific air resistance approx. $7 \cdot 10^4$ rayl $\times$ meter, compression 20 per cent under a static load of $2 \cdot 10^4$ N/m² ($\simeq$ 0.2 kg/cm²). The curves a and b have been measured with the new set-up without and with compensation respectively.

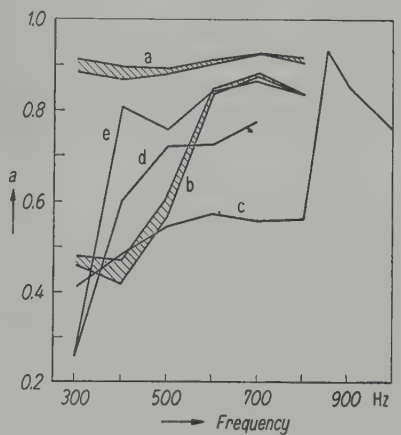


Fig. 5. As Fig. 4 with a highly flexible material (Kramfors, 3.5 cm, painted).
a) without compensation for leakage,
b) with compensation,
c) in interferometer ø = 9 cm,
d) in interferometer ø = 25.5 cm,
e) with compensation, no contact.

A sample cut out of the layer and measured in a 9 cm Ø interferometer yielded curve c. This curve deviates from what is theoretically expected for an incompressible layer of that thickness, from which may be inferred that obviously the great compressibility of the material is of major importance for its absorptive behaviour. But then a 9 cm interferometer should be considered too narrow for a reliable measurement. Therefore another sample was measured in an interferometer of 25.5 cm diameter (curve d). The curve has changed indeed and resembles the compensated curve reasonably. The remaining difference b,d may originate in the difference of hindrance of the vibration of the solid material. Curve b relates to a condition in which the hindrance was caused by the inner tube resting

*) I thank Mr. M. DE KRUIJFF for assistance in developing the apparatus and carrying out the measurements.

against the surface, curve d to one in which there was hindrance of the cylindrical surface of the sample by the wall of the interferometer. We finally attempted therefore to measure "in the field" with both cylinders withdrawn so far that the material was just untouched (curve e). A considerable improvement was found. The measured characteristic is now much more favourable than would be expected for a non-flexible layer that would otherwise have the same properties, in agreement with previously published theoretical predictions [4].

From these preliminary results we summarize as follows:

a) field measurements of normal absorption without a guard tube seem unreliable;

b) the new method with a guard tube seems reliable and may even be the only method available up at present for measuring rather flexible materials;

c) there has been found experimental evidence for the prediction made on theoretical grounds that flexibility of the solid material of a porous absorber improves the absorption at low frequencies.

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[2] Jones, E., Edelman. S. and London, A., J. Res. nat. Bur. Stand. 49 [1952], 17 ··· 20.

[3] a. Kosten, C. W., Building Research Congress 1951, Division 3, p. 32 (Princess Press Ltd., London);

b. Kosten, C. W., Technical aspects of sound (ed. E. G. Richardson), division I, chapter 4, p. 67 (Elsevier Publ. Co., Amsterdam 1953).

[4] See [3b], p. 79, eq. I, 49.

Discussion

Brüel, P. V.: 1) Have the inner and outer tube been subdivided in order to avoid non-plane modes of vibration. 2) Isn't it sufficient to compensate only by adjustment of the intensity while disregarding possible phase differences? This would constitute a considerable simplification and enable automatical adjustment of compensation.

Lecturer: 1) The measurements were only carried out at such low frequencies that progressive non-plane waves could not exist. In earlier experiments avoiding non-plane waves by subdivision has turned out more difficult than it seems at first sight. 2) Small mismatch of phase has lead to considerable errors; disregarding phase differences is intolerable in my opinion.

Narain, C.K.: How is sound leakage avoided when the measuring tubes are not in contact with the surface of the material?

Lecturer: Only by equalizing the sound pressures in both tubes.

Waterhouse, R. V.: What precautions were taken to seal the join between sample and tube before compensation?

Lecturer: None. "Sealing" is only done acoustically by equalizing the pressures.

Beranek, L. L.: I wish to support Dr. Kosten's statement that the clamping effect on small samples in impedance tubes is very important. During measurements on samples 0.2×0.2 m² we found serious effects due to clamping up to 400 c/s.

THE MEASUREMENT OF ACOUSTIC PRESSURES IN MOVING AIR STREAMS

by R. W. Leonard

University of California, Los Angeles

Summary. The self-noise of several windscreens has been obtained at velocities from 5 to 15 m/s and frequencies from 90 tot 3000 c/s. The study indicates an optimal screen flow resistance of 0.06 ϱc. Streamlining a circular cylinder may reduce the noise 3 dB at frequencies 300 to 1000 c/s. The spectrum level of the noise varies as the 6th power of the air velocity near 250 c/s and as the 8th power near 1000 c/s.

Sommaire. On a mesuré le bruit propre de plusieurs écrans à des vitesses comprises entre 5 et 15 m/s, et pour des fréquences de 90 à 3000 Hz. Les résultats de l'étude donnent 0,06 ϱc comme résistance optimum de l'écran à l'écoulement. Par profilage en cylindre aérodynamique, on peut réduire le bruit à 3 dB aux fréquences 300 à 1000 Hz. Le niveau spectral du bruit varie comme la puissance 6 de la vitesse de l'air, au voisinage de 250 Hz, et comme la puissance 8, près de 1000 Hz.

Zusammenfassung. Das Eigengeräusch verschiedener Mikrophon-Windschirme wurde im Frequenzbereich von 90 bis 3000 Hz bei Geschwindigkeiten von 5 bis 15 m/s ermittelt. Die Untersuchung ergab einen günstigsten Schirm-Strömungswiderstand von 0,06 ϱc. Ein stromlinienförmiger Zylinder kann das Geräusch im Bereich von 300 bis 1000 Hz um 3 dB herabsetzen. Der Geräuschpegel ändert sich im Spektralbereich um 250 Hz mit der 6. Potenz der Luftgeschwindigkeit und im Bereich um 1000 Hz mit der 8. Potenz.

1. Introduction

In making sound pressure measurements in moving airstreams, it is important to know the minimal acoustic pressures that can be measured in the presence of the self-noise of the microphone. The use of screens to shield microphones from air-flow is as old as the microphone itself, yet there seem to be little quantitative data in the literature. It is the purpose of this paper to present a limited amount of such data in the form of self-noise spectra showing the dependence of the spectrum level on air-stream velocity and frequency.

2. Apparatus

The apparatus consists of a rotating platform carrying a Western Electric 633A microphone on a tubular arm at a radius of 1.4 m from the axis of rotation. This rotating assembly is installed in an anechoic chamber; it is driven by a belt from a line shaft in the chamber. The variable-speed motor driving the line shaft is in the control room outside the anechoic chamber. The microphone connections are made through a telephone jack used as a rotating joint in the cable leading to the measuring equipment. The amplified signal from the microphone is passed through a half-octave filter set and read on a vacuum tube voltmeter. The half-octave levels are then reduced to spectrum levels for presentation. The period of rotation of the microphone arm is obtained by stop watch measurement.

The windscreens were formed out of rabbit wire (1 cm mesh) and covered with cloth of different specific flow resistance. The fabric on the larger cylindrical screens was taped to the surfaces at the edges with masking tape.

The cylinders, both circular and streamlined, moved through the air in a direction perpendicular to their axes. The ends of the cylinders were covered with cloth except where wood ends were used.

3. Measurements

A typical set of data is shown in the figure where spectrum level is plotted against frequency and air-velocity. These curves are for a cylindrical windscreen of diameter 12.5 cm, length 21 cm and covered with materials of different specific flow resistance. For a cylinder of these dimensions a specific flow resistance of 0.06 ϱc seems optimal at the lower frequencies (lines drawn in full). The thin dashed curve (0.03 ϱc) and the heavier dashed curve (0.25 ϱc) are tied to the corresponding solid curve by diagonal lines.

The velocities chosen are related to each other by multiples of the seventh root of ten. This makes it possible to estimate the power law in velocity. A

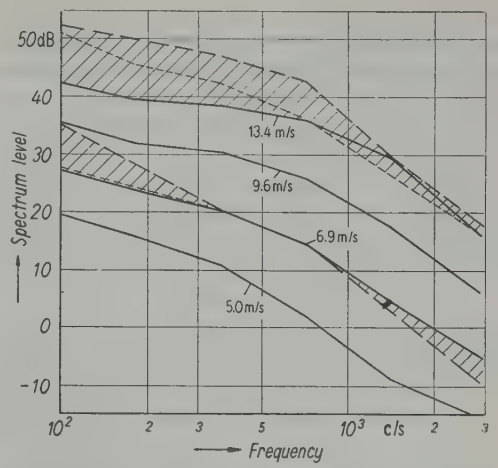


Fig. 1. Spectrum level in dB, reference 2×10^{-5} N/m², as a function of frequency and air stream velocity for a cloth-covered circular cylindrical screen of diameter 12.5 cm and length 21 cm. The air stream is set perpendicular to the axis of the cylinder. Solid curves for specific flow resistance of 0.06 ϱc. Thin dashed curve 0.03 ϱc. Heavier dashed curves 0.25 ϱc. The broken curves are linked to the solid curve of equal air velocity by diagonal lines.

10 dB spacing indicates the spectrum level (power per 1 c/s band) varying as the 7th power of the air-velocity. It is interesting to note that a 7th power dependence is exceeded at 500 c/s and an 8th power at 1000 c/s.

The spectrum levels of the 663A microphone without a screen are between 25 and 30 dB higher than those shown in the figure at the same air-speeds.

A streamlined screen of frontal area equal to the cylinder used for the measurements in the figure showed 3 to 5 dB less spectrum level for the same air-speed above 300 c/s. The streamlined shape was 12.5 cm thick and 28 cm in chord. The spectrum level of this screen was also minimised by a covering of 0.06 ϱc material for frequencies below 1000 c/s.

No systematic study of scale effect has been possible thus far but measurements have been made on a smaller circular cylinder (7 cm in diameter and 12 cm long) of the same proportions, and it was observed to be 6 to 8 dB noisier than the one used above. The 0.25 ϱc covering may or may not have been optimal for its dimensions.

In closing, the effect of turbulent flow requires mention. It was observed that the spectrum level was about 6 dB higher when the small cylinder was in the wake of an irregular body just upstream from the screen. The added noise due to the irregular body was estimated by reversing the direction of motion of the combination. Thus, it may be necessary to specify the turbulence in the air-stream

before it is possible to predict the self-noise spectrum of a windscreen microphone combination. In any event a much more extensive study is needed to put acoustic pressure measurements in airstreams on an accurate quantitatieve basis.

Discussion

ZWIKKER, C.: It seemed that the curves dB $= f(\omega)$ toward the low frequencies merged into a sequence corresponding with a fourth power law ($Ia v^4$) as predicted by the theory. Is there any reason to expect a higher exponent law at higher frequencies?

LECTURER: The lowest exponent measured is 5 but probably tends toward 4. No explanation is as yet found for the higher exponents.

MAWARDI, O. K.: In answer to the remark by Professor ZWIKKER, I would like to point out that simple poles cannot be used to represent noise of aerodynamic origin. High order poles, however, can and, as LIGHTHILL has shown, the acoustic power would then follow an eighth power law.

THE MEASUREMENT OF ACOUSTIC IMPEDANCE OF SMALL SAMPLES

by O. K. MAWARDI

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Summary. A review is presented of some of the better known methods of measurement of acoustic impedance. The results obtained by these methods are critically examined. The effect of several small-scale phenomena on the results is discussed. The outcome of a survey sponsored by the American Standards Association on the evaluation of normal acoustic impedance from small samples is described and a comparison is made with expected absorption coefficients from reverberation room measurements.

Sommaire. On passe en revue quelques unes des méthodes les plus connues de mesure de l'impédance acoustique. Examen critique des résultats fournis par ces méthodes. Influence de plusieurs petits phénomènes sur ces résultats. On indique les conclusions d'une étude entreprise sous l'égide de l'American Standards Association au sujet de la détermination de l'impédance acoustique normale de petites éprouvettes, et on fait la comparaison avec les coefficients d'absorption prévus d'après les résultats de mesures en salle réverbérante.

Zusammenfassung. Es wird ein Überblick über einige der bekanntesten Methoden zur Messung akustischer Impedanzen gegeben und dann werden die Resultate dieser Meßmethoden kritisch betrachtet. Die Auswirkung einiger Phänomene geringerer Bedeutung auf die Resultate wird diskutiert. Das Ergebnis einer mit Unterstützung der American Standards Association durchgeführten Untersuchung über die Bestimmung der akustischen Impedanz kleiner Proben wird beschrieben und mit den nach Hallraum-Messungen zu erwartenden Absorptionskoeffizienten verglichen.

1. Introduction

Several methods of measurement of acoustic impedance can be traced in the scientific literature. Quite a few of those methods are best suited for the determination of the impedance at normal incidence from small samples. These methods are of primary importance to the manufacturers of acoustic materials since they determine the inherent acoustic properties of the material. It is true that the evaluation of the behaviour of acoustic materials on a large scale is also an essential question, but it is apparent that the behaviour in the latter state is influenced to a great extent by factors extraneous to the material (method of fixation, etc.).

An important question in connection with the evaluation of impedance is the accuracy involved in the measurements. In order to develop a reliable method for the evaluation and rating of the absorptive properties of acoustic materials, the American Society for Testing Materials together with the American Standards Association coordinated its efforts in conducting a series of round-robin tests. These tests were made in order to evaluate the degree of agreement between measurements in different laboratories. The purpose of this paper is to present the outcome of this survey and also to examine critically the limitations commonly encountered in the methods of measurement. It is intended here to make some remarks of a general nature rather than to discuss each known method on an individual basis.

2. Remarks on methods of measurements

In order to simplify the instrumentation aspect of the problem the normal specific acoustic impedance for a material is usually evaluated for plane waves. Upon consideration of the known methods of impedance measurement one can classify without too much difficulty nearly all of the methods in two broad groups. In the first are those which utilize data measured at the surface of the material. In the second group, on the other hand, measure-

ments are made at points remote from the surface of the sample.

Of the two groups, the first seems to be more direct and one would believe that it would yield more readily the value of the normal specific impedance. But in fact, this is not true in general. Several disturbing factors set by the physics of the situation causes it to depart from the idealized condition of a plane wave incident on the material under test. For instance, let it be assumed that the surface irregularity of the material (Fig. 1a) be of the order of a depth l, or let a detector whose size

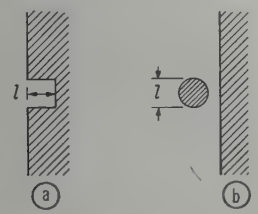


Fig. 1. Two idealized configurations for the irregularities mentioned in the first group of methods of measurement.

is described by a characteristic length l be introduced in the vicinity of the material (Fig. 1b). It is readily shown that the uncertainty in the value of the measured impedance is of the same order of magnitude as that for the time that is derived from an "uncertainty principle"

$$\Delta t \cdot f \simeq 1,$$

where $\Delta t \simeq l/c$, c being the velocity of sound, f the frequency and l defined from Fig. 1. By means of the relation $f = c/\lambda$, the above becomes

$$l \cdot 1/\lambda \simeq 1.$$

The truth of the previous statement is easily checked on Fig. 1a. At the mouth of the irregularity the impedance is approximately that of spherical waves, while away from it, it is practically that of plane waves. Roughly then, the two local impedances differ by kl, i.e. approximately l/λ which proves the point.

On the other hand, when the methods of the second group are considered, it is expected that local field irregularities, if any, will have decayed at a reasonably short distance from the sample. A typical example of such an instance is encountered in any of the techniques involving standing wave patterns in a tube. Slight consideration of the matter on hand leads to the conclusion that the measured value of the impedance is not the true specific impedance, but an *equivalent* impedance which will yield the correct values for the energy dissipated in the sample and for the reactive energy capable of being stored in the material.

The limitation on this second group of methods of measurements is often turned to advantage in the evaluation of the dependence of the impedance on edge effects (diffraction effects) or absorption cross-section, etc. Typical schematic arrangements are shown in Fig. 2a and 2b. The impedance as evaluated

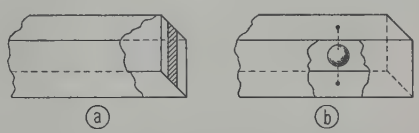


Fig. 2. Possible experimental arrangement for the evaluation of edge effects (a) and scattering cross-section (b).

by means of the second group is usually found from Q, the quality factor, of this system or from energy considerations. It is not difficult to demonstrate that nearly all methods can indeed be reduced to a formulation leading to either one of the two above mentioned concepts.

Due to the wide variety of the methods that have been developed it is difficult to conceive a simple and general method of comparison of relative merits.

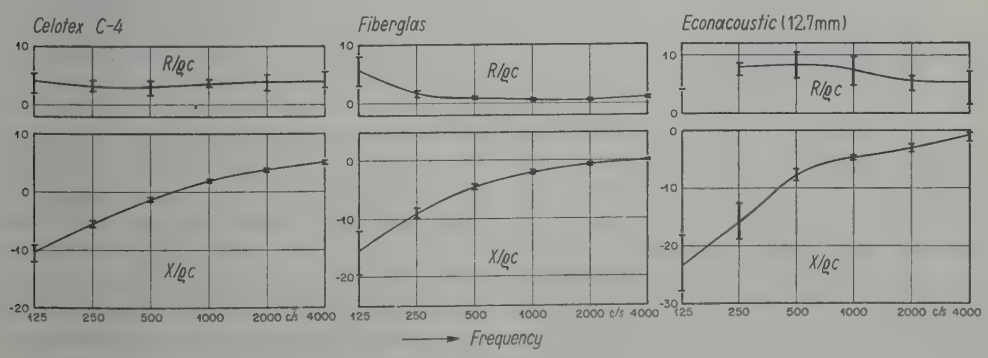


Fig. 3. Experimental findings of round-robin test described in the text. $R/\varrho c$ values at the lowest frequency in Fig. 3c are not indicated, as long tube measurements become unreliable for hard specimens. The point shown for 125 c/s corresponds to short tube finding.

3. The data

It is now intended to present typical results of measurements as illustrated on the round-robin test previously mentioned in the introduction. The samples sent to each of the participants were chosen as representative of commercial tiles and were carefully selected on the basis of reverberation measurements. At 5000 c/s the samples indicated a maximum deviation of the absorption coefficient of 0.01 from the mean. All measurements were to correspond to the case of *rigid* backing. The accompanying graphs (Fig. 3) illustrate the results obtained from different laboratories using so-called "long tube or short tube methods."

4. Conclusion

Inspection of the data indicates that the scatter in it is not too serious. Furthermore, it compares very favourably by contrast with the scatter usually encountered in the "observed" absorption coefficients. By observed coefficient is meant the value deduced from the conventional reverberation formula with no correction for the diffraction effect of the edges. On the basis of the result of the survey it seems reasonable to suggest the use of the impedance in specifying acoustic material.

The author would like to express his gratitude to Mr. H. C. Lang for making the results of the survey available to him.

Discussion

Vogel, Th. M.: M. Mawardi a fait des remarques très importantes sur le rôle des irrégularités du matériau. Je voudrais rappeler que j'ai décrit récemment [1] une méthode de mesure d'où l'on peut déduire les impédances normales et qui diminue l'importance des irregularités: en effet, d'une part les échantillons ont 1 m² de surface, et de l'autre on peut montrer que c'est la moyenne superficielle de α que l'on mesure.

[1] Vogel, Th. M., Ann. Télécomm. 8 [1953], 93.

MEASUREMENT OF FLOW VELOCITIES IN WIND TUNNELS BY ACOUSTIC METHODS *)

by O. K. Mawardi

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Summary. An acoustic method is proposed for the evaluation of the average flow velocity in wind tunnels. The method is based on an estimate of the effect of a mean velocity of the medium on the time of arrival of a sound signal. The effect of the boundary layer on the propagation of sound waves is discussed. The results obtained by the method are expected to yield a reasonably good accuracy.

Sommaire. On propose une méthode acoustique de détermination de la vitesse moyenne de l'écoulement dans les souffleries. Dans cette méthode, on évalue l'influence d'une vitesse moyenne du milieu sur le temps de parcours d'un signal sonore. On examine l'effet de la couche limite sur la propagation des ondes sonores. On estime que cette méthode doit fournir des résultats avec une précision raisonnable.

Zusammenfassung. Es wird eine akustische Methode vorgeschlagen, um die durchschnittlichen Flußgeschwindigkeiten in Windkanälen zu bestimmen. Bei dieser Methode benutzt man den Einfluß einer Bewegung des Mediums auf die Ankunftszeit eines Schallsignals. Die Auswirkung der Grenzschicht auf die Ausbreitung von Schallwellen wird diskutiert. Man erwartet, daß die mit dieser Methode gewonnenen Ergebnisse eine ziemlich gute Genauigkeit ergeben.

1. Introduction

In conventional methods of measurement of flow velocities an instrument such as a pitot tube or a plate thermometer is usually inserted inside the main stream. The stream velocity is then evaluated in terms of the flow parameters registered by the instrument. An unavoidable error in the results, however, is introduced when using such methods, since the presence of the instruments in the main flow locally disturbs it.

In order to avoid the introduction of instruments in the flow several investigators [1] have tried acoustical techniques for the evaluation of flow velocities. When boundary layer effects are not taken into account, these methods yield incorrect values for the velocities. As boundary layer corrections are difficult to evaluate, these methods have been abandoned. The technique proposed here, on the other hand, makes use of propagation in the boundary layer and a preliminary approximate theory of propagation of sound waves in the boundary layer is presented.

*) This research has been supported in part by a contract with the United States Air Force.

The advantage of this method is that under certain conditions the disturbances introduced in the main stream are considerably reduced.

2. Propagation in the boundary layer

The concept of the boundary layer was introduced in 1904 by PRANDTL [2]. This layer is a region of comparatively small thickness near the surface of a body moving with respect to the fluid around it. In that layer the viscous forces are comparable to the inertia forces and the relative speed between the body and the fluid decreases rapidly to zero.

This thin layer is an ideal sound channel for the purpose on hand. When the main stream velocity is supersonic the whole sound energy would be localized in the boundary layer. Unfortunately, this state of affairs is rarely encountered. As is well known, the flow in the boundary layer is highly unstable and extensive investigations have been done to study the stability criteria of flow in boundary layers. In the approximate analysis given here it will be assumed that the flow is laminar and that conditions are such that the boundary layer is stable.

The equations of flow in the boundary layer will be now established. The discussion will assume a two dimensional viscous flow along a flat plate. Following the conventional derivations for flow in a boundary layer [3], one starts from the equations of motion and continuity. These are,

$$\left.\begin{aligned} \frac{\partial u}{\partial t} + u\frac{\partial u}{\partial x} + v\frac{\partial u}{\partial y} &= -\frac{1}{\varrho}\frac{\partial p}{\partial x} + \nu\nabla^2 u \\ \frac{\partial v}{\partial t} + u\frac{\partial v}{\partial x} + v\frac{\partial v}{\partial y} &= -\frac{1}{\varrho}\frac{\partial p}{\partial y} + \nu\nabla^2 v \end{aligned}\right\} \quad (1)$$

$$\frac{\partial u}{\partial x} + \frac{\partial v}{\partial y} = 0 \quad (2)$$

where u, v are the component velocities, ϱ the density, p the pressure and ν the kinematic viscosity. The conventional assumption of incompressibility has also been made (eq. 2). This is justified because of the small size of the boundary layer thickness. Let the velocity u be $(u^* + U)$, in which U, the main stream velocity, is a function of y only, the y-axis being perpendicular to the boundary. For infinitesimal acoustic disturbances superimposed on the motion in the boundary layer, all quantities of second order can be neglected. Let it also be assumed that the v component of velocity is dependent on the form,

$$v = \Phi(y)e^{jkx}. \quad (3)$$

By virtue of eq. (2), the relation for u^* is found to satisfy the condition

$$jku^* = -\partial v/\partial y.$$

For a time dependence $e^{j\omega t}$ elimination of the pressure between the two relations (1) and use of (2) and (3) leads to,

$$\left(U - \frac{\omega}{k}\right)(\Phi'' - k^2\Phi) - U''\Phi = \frac{-j}{k\nu}(\Phi'''' + 2k^2\Phi'' + k^4\Phi) \quad (4)$$

which is the well-known relation for the boundary layer near a flat plate [4]. Let it be now assumed that the distribution of the velocity $U(y)$ is as shown in Fig. 1, b_1 being the thickness of the boundary

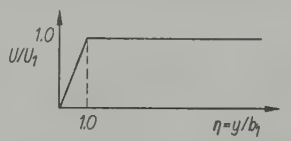


Fig. 1. Velocity distribution in the boundary layer.

layer. This departs from the conventional BLASIUS distribution, but the linear assumption has been adopted since it simplifies the mathematical manipulations. Writing

$$U = U_1\frac{y}{b_1}; \quad \frac{y}{b_1} = \eta; \quad kb_1 = \lambda,$$

$$\frac{\omega b_1}{U_1} = \omega^* \text{ and } \frac{Ub_1}{\nu} = \mathbf{R},$$

the above equation (4) becomes,

$$\left(\eta - \frac{\omega^*}{\lambda}\right)[\Phi''(\eta) - \lambda^2\Phi(\eta)] = \frac{-j}{k\mathbf{R}}(\Phi'''' + 2k^2\Phi'' + k^4\Phi). \quad (4')$$

For high REYNOLDS numbers, which is the case of most interest to us (4') transforms to

$$(\eta - \omega^*/\lambda)(\Phi'' - \lambda^2\Phi) = 0. \quad (5)$$

As long as $(\eta - \omega^*/\lambda) \neq 0$, the equation to discuss is

$$\Phi'' - \lambda^2\Phi = 0. \quad (6)$$

The solution of eq. (6) is

$$\Phi = Ae^{\lambda\eta} + Be^{-\lambda\eta}. \quad (7)$$

Introducing the boundary condition requiring v to vanish at $\eta = 0$, there follows

$$\Phi = \sinh\lambda\eta. \quad (8)$$

Integration of the left hand side of eq. (4), with the proviso of $\mathbf{R} \gg 1$, leads to [5]

$$\left(1 - \frac{\omega^*}{\lambda}\right)\Delta[\Phi'(\eta)] - \Delta\left(\frac{\partial U}{\partial\eta}\right)\Phi = 0 \quad (9)$$

where Δ refers to a finite difference and where the integration has been carried between the two sides of the line $\eta = 1$, i.e., from the boundary layer to the mean stream.

As the model chosen implies that Φ is a function of y inside the boundary layer and is zero outside of the layer, i.e., in the free stream, it is possible to write

$$\Delta[\Phi'(\eta)] = \lambda\cosh\lambda.$$

Also by inspection of Fig. 1, $\Delta(\partial U/\partial\eta) = 1$. Con-

sequently substitution in the above relation (9) leads to

$$\left(1 - \frac{\omega^*}{\lambda}\right) \lambda \cosh \lambda - \sinh \lambda = 0 \qquad (10)$$

which is the relation between the forcing frequency and the wave number. The functional relation between $\omega^* = \omega b_1/U_1$ and $\lambda = kb_1$ is shown in graphical form in Fig. 2.

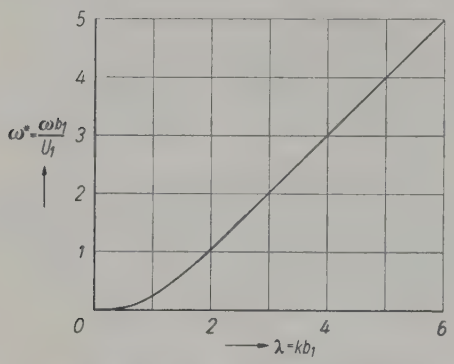


Fig. 2. Functional dependence of the relative wave number $\lambda = kb_1$ on the relative frequency $\omega^* = \omega b_1/U_1$.

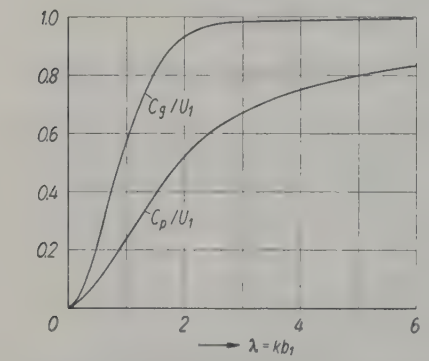


Fig. 3. Dependence of the group velocity C_g and the phase velocity C_p on the relative number $\lambda = kb_1$.

It is now possible to discuss the propagation of sound waves in the boundary layer by means of eq. (10) since it yields both the group and phase velocities (Fig. 3). These two velocities are

$$C_{phase} = \frac{\omega}{k} = U_1 \left(\frac{\omega^*}{\lambda}\right) = U_1 \left(1 - \frac{\tanh \lambda}{\lambda}\right)$$

$$C_{group} = \frac{d\omega}{dk} = U_1 (1 - \operatorname{sech}^2 \lambda).$$

3. The experimental procedure

Two possible arrangements suggested by the previous discussion can be used. A source S sends a signal which is picked up by two microphones placed at A and B respectively (Fig. 4), and at a distance l apart. When the signal sent by S is steady

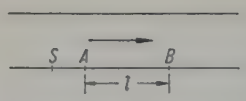


Fig. 4. Proposed experimental arrangement; S is source position, A and B are the microphone locations.

and has a time dependence of the form $\cos \omega t$, then the phase difference θ between the signals picked up at A and B is related to the main stream velocity U_1 in the manner,

$$\theta = \frac{\omega l}{U_1 \left(1 - \frac{\tanh kb_1}{kb_1}\right)}$$

from which the velocity U_1 is determined. On the other hand when the emitted signal has a time dependence $\cos \omega_c t \ (1 + \cos \omega_m t)$ then the phase difference of the *envelope* is related to the stream velocity U_1 through the expression

$$\theta = \frac{\omega_m l}{U_1 (1 - \operatorname{sech}^2 kb_1)}.$$

Since the group velocity in the latter expression very rapidly reaches its maximum value, it is

$$\theta = \frac{\omega_m l}{U_1},$$

indicating that the second possibility is more convenient.

References

[1] Eisenstein, J., Clark, K., and Carlson, F., U.S. Scientific Research and Development Report No. 5369.
[2] Dryden, H. L., Advances in Applied Mechanics. Academic Press Inc., 1948, Vol. I, p. 2 ff.
[3] Goldstein, S., Modern developments in fluid dynamics. Oxford University Press 1 [1950], 116.
[4] Goldstein, S., Modern developments in fluid dynamics. Oxford University Press 1 [1950], 197.
[5] Lord Rayleigh, Scientific Papers. Cambridge University Press 3 [1902], 17.

Discussion

Richardson, E. G.: In view of the fact that the source will radiate from the walls of the tunnel in all directions into the boundary layer in which the velocity is diminishing rapidly, will not various waves refracted along different paths arrive at the microphone, and will there not be difficulty in distinguishing them, particularly as they will be differently damped?

Lecturer: At supersonic speeds all the sound is restricted to the boundary layer. We have further calculated the zone of silence in the layer and found that the first skip distance is of the order of about eight times the thickness of the layer. Since the boundary layer thickness is nearly 5×10^{-4} m, it is safe to neglect completely the interference effects and consider the boundary layer as an ideal propagation channel.

Pahud, J. Ph.: Comment sont construits les transducteurs émetteurs et récepteurs?

Lecturer: Les transducteurs sont des microphones condensateurs.

A MEASURING SET FOR ACOUSTICAL AND MECHANICAL IMPEDANCES

by J. Y. Morton
Post Office Engineering Department, London.

Summary. The method used is to compare the impedances to be measured with known impedances of the same type. The impedances are applied to electro-acoustical or electro-mechanical transducers which adapt a common electrical circuit for the type of measurement required. Facilities are provided for giving readings of the modulus and phase of the unknown impedance.

Sommaire. Dans la méthode utilisée, on compare les impédances à mesurer avec des impédances connues, du même type. Les impédances sont branchées sur des transducteurs électro-acoustiques ou électro-mécaniques, qui assurent l'adaptation à un circuit électrique ordinaire, correspondant au type de mesure à faire. Il est prévu des dispositifs permettant de connaître le module et la phase de l'impédance inconnue.

Zusammenfassung. Bei der beschriebenen Meßmethode werden die zu messenden Impedanzen mit bekannten Impedanzen gleichen Typs verglichen. Die Systeme werden über elektroakustische oder elektromechanische Wandler an einen gemeinsamen elektrischen Kreis angepaßt, welcher der gerade erforderlichen Messung entspricht. Es sind Möglichkeiten vorgesehen, um eine direkte Anzeige von Betrag und Phase der unbekannten Impedanz zu erhalten.

1. Introduction

The equipment was designed as a research tool to measure the acoustical load impedance of hearing aid air-conduction receivers and the mechanical load impedance of bone-conduction receivers in the frequency range 200 to 5000 c/s with the object of providing design information for an artificial ear and an artificial mastoid, but as will be appreciated it has a far wider field of use. The measurement of acoustical impedance by a similar method has been demonstrated by Mr. E. R. Wigan of the British Signals Research and Development Establishment; this measuring set is a further development, and enables either a mechanical or an acoustical impedance to be measured.

2. Method

The method is best illustrated by analogous electrical circuits and it will be seen from these that the design can also cope with measurements of electrical impedance.

2.1 *Constant voltage*

In Fig. 1 a generator of e.m.f. E_0 and internal impedance Z_0 has a pair of output terminals to which various impedances may be connected, and between which a potential difference V exists. If the terminals are open-circuited the terminal potential difference is V_0 and is equal to E_0 (Fig. 1a). If now an unknown impedance Z_x is connected to the terminals the voltage changes but it may be restored to its original value V_0, in magnitude and phase, by increasing the generator e.m.f. by some amount E_1 as shown in Fig. 1b. Similarly for a known impedance Z_k (Fig. 1c) a corresponding e.m.f. E_2 may be found. The solution of the three simple equations for the three parts of Fig. 1 gives

$$Z_x = Z_k \frac{E_2}{E_1}.$$

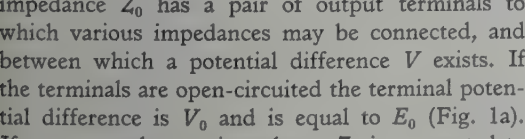


Fig. 1. Constant voltage circuit.

2.1.1. Acoustical analogue

In Fig. 2a the generator of Fig. 1a is replaced by a sound source including and terminated in a small cavity with an opening defining a measuring plane which corresponds with the generator output terminals; ${}_aZ_0$ is the acoustical impedance of this source. The generator e.m.f. E_0 of Fig. 1a is replaced by the sound pressure P_0 and the terminal voltage V_0 by the sound pressure p_0 measured at the measuring plane when no volume current is permitted to flow, i.e. when the aperture is closed by a rigid wall.

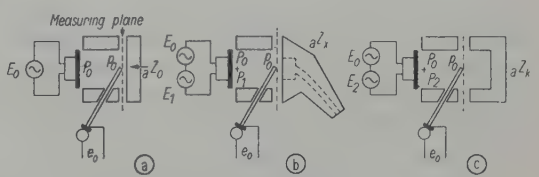


Fig. 2. Acoustical analogue of Fig. 1.

If an unknown acoustical impedance ${}_aZ_x$ is presented externally to the measuring plane, Fig. 2b, the sound pressure measured there, p say, will in general differ from the blocked value p_0. The input from the source may be altered to restore p to its original value p_0 in magnitude and phase. The change in input may be specified by the resultant increase P_1, in sound pressure, measured with the

aperture blocked. Clearly P_1 is the analogue of E_1 in Fig. 1b. Similarly if a known impedance ${}_aZ_k$ is presented at the measuring plane, Fig. 2c, a sound pressure P_2 may be found which is the analogue of E_2 of fig. 1c. By analogy with the electrical circuits ${}_aZ_x = {}_aZ_k\, P_2/P_1$.

It is convenient to use acoustical compliances as known impedances.

2.1.2. Adaptation to electrical measurements

The measuring plane pressure p_0 may be converted into an electrical e.m.f. e_0 by means of a probe tube microphone, and the sound pressure increments P_1 and P_2 derived from electrical increments E_1 and E_2 by using a linear electro-acoustical transducer. In terms of the input voltage to this transducer ${}_aZ_x = {}_aZ_k\, E_2/E_1$.

The probe tube microphone is part of the source impedance ${}_aZ_0$ and need not be a linear transducer.

2.2. *Constant current*

In Fig. 3a the terminal current delivered by the generator has the value $I_0 = E_0/Z_0$ when the terminals are short-circuited. If impedances Z_x and Z_k are connected as in Figs 3b and 3c, the output

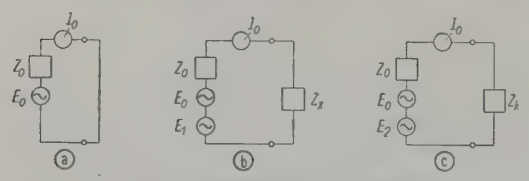


Fig. 3. Constant current circuit.

current changes, but may be restored to I_0, in magnitude and phase, by injecting additional e.m.f.'s E_1 and E_2 respectively. The solution of the equations for the three separate conditions gives $Z_x = Z_k\, E_1/E_2$.

2.2.1. Mechanical analogue

Fig. 4 represents the mechanical analogue of Fig. 3; the velocity of the measuring surface v_0 represents the constant current I_0 and is measured by a piezo-electric element. When an unknown impedance

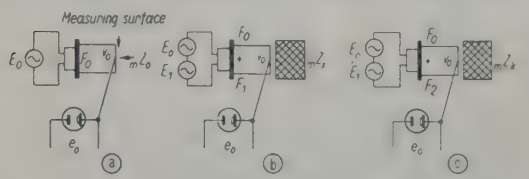


Fig. 4. Acoustical analogue of Fig. 3.

${}_mZ_x$ is applied to the measuring surface an increment of F_1 in the force F_0 applied to it is required to restore its velocity to its original value, Fig. 4b. Similarly an increment of F_2 is required when a know impendance ${}_mZ_k$, which takes the form of a mass, is applied, Fig. 4c. By analogy with the electrical circuit ${}_mZ_x = {}_mZ_k\, F_1/F_2$.

In terms of input voltage to the linear electro-mechanical transducer producing F_1 and F_2, ${}_mZ_x = {}_mZ_k\, E_1/E_2$.

The piezo-electric element is part of the source impedance ${}_mZ_0$ and need not be a linear transducer.

3. Electrical circuitry and operating procedure

Fig. 5 gives one of the many possible circuits which may be used. The simplest method of explaining the function of the various circuit elements is to go through the sequence of operations necessary to make a measurement. In the first stage E_0 is applied to the electro-mechanical transducer through the combining unit. The velocity v_0 of the measuring surface causes an e.m.f. e_0 to be generated in the piezoelectric element attached to it. This e.m.f. is compared with an auxiliary source e.m.f. e_0' which is adjusted in magnitude and phase until it equals e_0 as shown in the null detector.

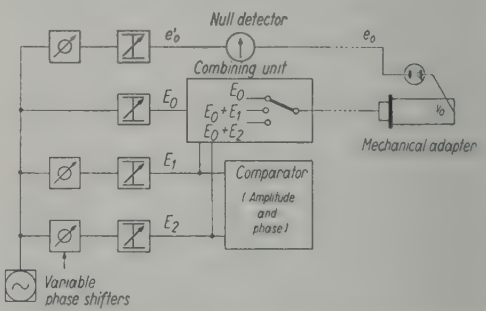


Fig. 5. Common electrical circuit and mechanical adapter.

In the second stage the unknown impedance is applied to the measuring surface and the E_1 source is adjusted in magnitude and phase until the null detector indicates that the velocity of the surface has been restored to v_0.

In the third stage a known mass is applied and the E_2 source is adjusted in magnitude and phase until the velocity has again been restored to v_0.

The comparator, which consists of two voltage dividers and a phase measuring set, is then brought into operation and gives the modulus and phase angle of the unknown impedance. The calibrations of the E_1 and E_2 phase shifters and level controls may be used to give an approximate value without recourse to the comparator.

In measuring acoustical impedances the mechanical adapter is replaced by an acoustical one and a similar procedure is applied. Impedances can be determined within $\pm 1\%$ if suitably designed electro-acoustical or electro-mechanical transducers and reference impedances are available.

4. Awkward position technique *)

A technique has been developed for measuring acoustical impedances which cannot normally be connected directly to the measuring plane, such as the impedance ${}_aZ_e$ looking into the ear from the plane of the tip of an ear-mould (Fig. 6a).

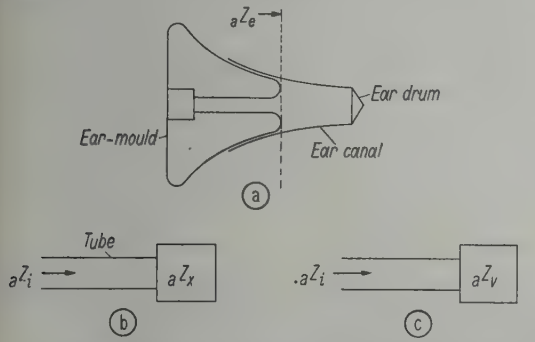


Fig. 6. Measurement of acoustical impedances in awkward positions.

The input impedance of a tube terminated by the unknown impedance ${}_aZ_x$ is measured (Fig. 6b). The tube is withdrawn from the acoustical system in which the unknown impedance is situated and the latter is replaced by a continuously variable acoustical impedance ${}_aZ_v$ (Fig. 6c) which is described below. The variable impedance is adjusted until the input impedance of the tube is repeated. This occurs when the variable impedance equals the unknown impedance, hence by measuring the former it is possible to obtain the value of the unknown impedance.

When measuring the impedance from the tips of ear-moulds the tube takes the form of the ear-mould canal.

4.1. *Continuously variable acoustical impedance*

Fig. 7a gives the electrical analogue. If the potential difference across the connections between the two electrical circuits is V and a current I flows the effective impedance of circuit No. 2 is given by $Z = V/I$. If circuit No. 2 is replaced by a generator, the e.m.f. of which is adjusted until its terminal voltage is equal to V (Fig. 7b) then the effective impedance of the generator will equal V/I which is equal to that of circuit No. 2. This follows from the fact that the generator and circuit No. 2 can both be replaced by a second generator of zero internal impedance and e.m.f. equal to V in accordance with the compensation theorem. As circuit No. 2 can have an infinite number of impedance values, the generator of Fig. 7b becomes a continuously variable electrical impedance.

*) This section formed part of the discussion, but is added to the paper for convenience of the reader.

The acoustical analogue of Figs. 7a and b is given in Figs. 7c and d. The sound pressure P in the connection between the two acoustical systems of Fig. 7c is the analogue of the voltage V in Fig. 7a. If system No. 2 is replaced by a sound source, P will in general change but it can be restored to its original value by adjusting the electrical stimulus to the source in magnitude and phase (Fig. 7d). By analogy with the electrical circuits the effective impedance of the sound source will then be the same as system No. 2, and as system No. 2 can have an infinite number of impedance values the sound source becomes a continuously variable acoustical impedance.

When measuring the effective impedance of the sound source it is necessary to maintain the sound pressure P at the plane of its connection to the measuring set. This is facilitated by using a probe tube microphone, a null detector and reference voltage source similar to that used in the impedance measuring set, and adjusting the E_1 channel of the measuring set (Fig. 5) in magnitude and phase until a null is obtained.

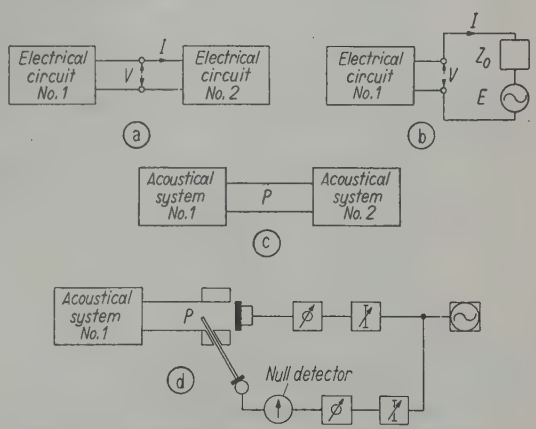


Fig. 7. Continuously variable acoustical impedance.

The sound source may be adjusted to a given acoustical impedance by connecting it to the measuring set and adjusting its electrical stimulus until the readings on the set correspond with the desired impedance. If this impedance is to be placed in another acoustical system it will again be necessary to maintain the sound pressure at the plane of connection of the sound source.

4.2. *Mechanical analogues*

The "awkward position technique" and a continuously variable mechanical impedance will have applications in the mechanical field if they can be achieved in practice, in particular the latter might be used to produce pure mechanical resistance or compliance.

Discussion

GAYFORD, M. L.: What are the limitations on accuracy of measurement at frequencies above 5 kc/s for acoustical impedance measurements?

LECTURER: Mainly the size of the microphone probe compared with the diameter of the measuring cavity and the volume of the measuring cavity.

BROWN, R. E. C.: What is the order of accuracy of the measurements obtained and the time taken to make the measurements?

LECTURER: For inert impedances the accuracy can be of the order of $\pm$ 1% and the time taken for individual measurements is of the order of 30 seconds. When measuring the mechanical impedance of people's mastoids and the acoustical impedance of their ears considerable time is required to position the subjects correctly and so it is only possible to make approximately 40 measurements a day. Repeat measurements show a scatter because of slight differences in the position of the subject and, in addition in the acoustical case, because of the difficulty of measuring through the low reactance of the ear-mould well cavity at high frequencies.

DADSON, R. S.: At the National Physical Laboratory we have independently been making measurements of the mechanical impedance of the mastoid process in connection with the objective calibration of bone-conduction receivers for audiometers and hearing aids. We have used a method equivalent to that described by Mr. MORTON and have found the method very useful and accurate. The results are given in a paper in section IV of the Congress. The frequency range covered is from about 100 to above 6000 c/s.

THE DESIGN OF AN ACOUSTIC IMPEDANCE METER

by A. K. NIELSEN

Telefon Fabrik Automatic, Copenhagen

Summary. The sound field in a cylindrical tube has been computed for various source configurations. The pressure distribution functions corresponding to simple piston sources are shown graphically, and the computations are utilised in the design of an acoustic impedance meter. Some results of impedance measurements are given.

Sommaire. On a calculé le champ sonore produit dans un tube cylindrique par des sources de différentes configurations. On donne sous forme graphique les fonctions distribution de la pression qui correspondent à des sources du type piston simple, et on se sert des résultats du calcul pour établir un impédancemètre acoustique. On donne quelques résultats de mesures d'impédance.

Zusammenfassung. Das Schallfeld in einem zylindrischen Rohr wurde für verschiedene Anordnungen der Schallquellen berechnet. Die Funktion der Schalldruckverteilung unter Voraussetzung einfacher Kolbenschwinger als Schallquellen wurde graphisch dargestellt. Die Berechnungen werden zur Konstruktion eines akustischen Impedanzmessers benutzt, und einige Resultate von Impedanzmessungen werden angegeben.

1. Introduction

The surface method [1], [2], [3] for the measurement of acoustic impedance is based on a sound source with a high intrinsic impedance producing a constant volume velocity into a short tube closed by the sample to be measured. The average sound pressure in the coupling tube is then a measure of the acoustic impedance of the sample.

A detailed knowledge of the sound field in the coupling tube is useful, and the first part of this paper is reserved for a general numerical discussion of the sound field due to various source configurations. In the second part of the paper it is shown how these computations may be utilised in the measurement of acoustic impedance.

List of symbols

a = tube radius
p = sound pressure
u = particle velocity
ϱ = density of medium
c = velocity of sound
$k = \omega/c$ = wave number
$\varkappa_n$ = n'th root of $J_1(x)$
$k_n = \varkappa_n/a$
$\mu_n^2 = k_n^2 - k^2$
Z_{tn} = specific acoustic impedance ratio of terminal for n'th order mode
$Z_{cn} = jk/\mu_n$ = characteristic specific impedance ratio of n'th order mode
$F\left(\frac{r}{a}, \frac{x}{a}\right)$ = pressure distribution function, r and x being cylindrical coordinates of points in the tube, r being measured from 0 to a, x from 0 to l.

2. The sound field in a tube of circular cross-section

A certain source distribution is assumed for $x = 0$ which may be expanded in a FOURIER-BESSEL series [4], [5],

$$u = \sum_{n=0}^{\infty} u_n J_0(k_n r)\, e^{j\omega t}; \qquad (1)$$

it is convenient to normalize the expansion so that $u_0 = 1$.

The sound pressure may be written as

$$p = \sum_{n=0}^{\infty} Z'_n(x)\, u_n\, J_0(k_n r)\, e^{j\omega t} = \varrho c\, (p_0 + \Sigma p_n)\, e^{j\omega t} \quad (2)$$

and the transfer impedance $Z_n'(x)$ is

$$Z_n'(x) = \varrho c\, Z_{cn}\, \frac{Z_{tn} \cosh \mu_n (l-x) + Z_{cn} \sinh \mu_n (l-x)}{Z_{tn} \sinh \mu_n l + Z_{cn} \cosh \mu_n l} \quad (3)$$

The pressure distribution function is now defined as

$$F\left(\frac{r}{a}, \frac{x}{a}\right) = \frac{\Sigma p_n}{jka}. \quad (4)$$

For low frequencies and a reasonable length of the tube, i.e. for $k \ll k_1$ and $\mu_1 (l-x) \gg 1$, $F(r/a, x/a)$ may be shown to be dependent on the geometry of the source only, viz.,

$$F\left(\frac{r}{a}, \frac{x}{a}\right) = \sum_{n=1}^{\infty} \varkappa_n^{-1}\, e^{-k_n x}\, u_n\, J_0(k_n r)$$

For a piston source extending from $r = 0$ to $r = a_1$ we find

$$u(r) = \left(\frac{a}{a_1}\right)^2, \quad 0 \leq r \leq a_1; \quad u(r) = 0, \quad a_1 < r \leq a$$

$$u(r) = 1 + \sum_{n=1}^{\infty} \frac{a}{a_1} \frac{2 J_1(k_n a_1)}{\varkappa_n J_0^2(\varkappa_n)} J_0(k_n r). \quad (5)$$

The corresponding $F(r/a, x/a)$ functions have been computed for some values of the parameters, Figs. 1 ··· 4.

3. The method of measuring acoustic impedance

It is at once evident from the curves Figs. 1 ··· 4 that for each particular sound source a radial distance may be found which is best suited for the measurement of the average sound pressure in each cross-section. Even a very irregular pressure distribution may be tolerated in a cross-section if the average pressure is measured correctly.

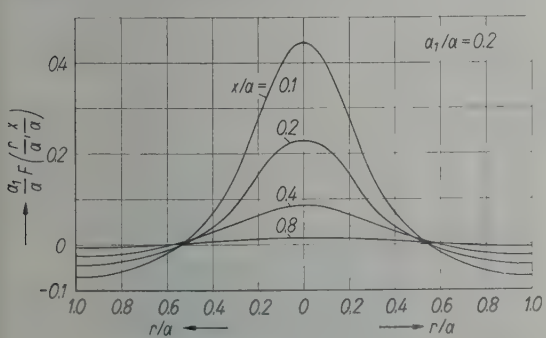


Fig. 1. Pressure distribution function for $a_1/a = 0.2$.

It was desirable to use condenser microphones both as a sound source and as a microphone. The "Western Electric" 640 AA could be used for both purposes, and the method used might be thought of as a variation of the usual pressure calibration by the reciprocity principle [6]. The usual reciprocity calibration is based on the knowledge of an acoustic impedance, e.g. that of a small cavity, from which the response of the microphones is deduced. Here, conversely, the responses of the microphones are known, but the impedance of the coupler should be determined.

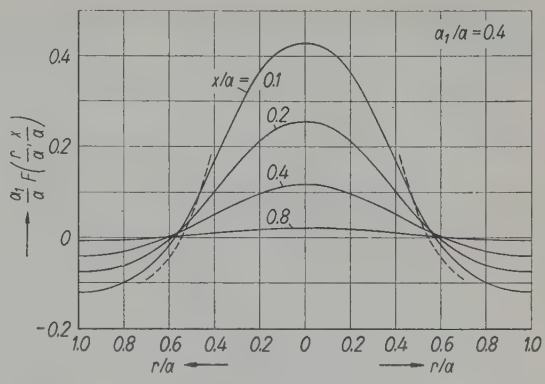


Fig. 2. Pressure distribution function for $a_1/a = 0.4$.

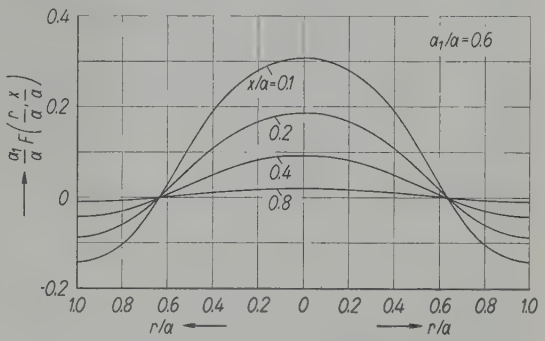


Fig. 3. Pressure distribution function for $a_1/a = 0.6$.

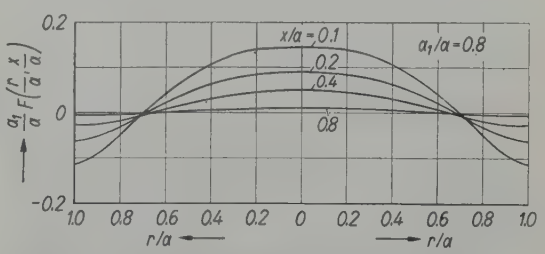


Fig. 4. Pressure distribution function for $a_1/a = 0.8$.

The principle of the apparatus is shown on Fig. 5. The tube diameter is 4.75 cm, the source microphone is mounted coaxially with the tube and the receiving microphone is placed close to the source, the sound pressure being measured through a very short tube. It is possible in this way to obtain a

reasonable flat frequency response up to 4 kc/s at least [3].

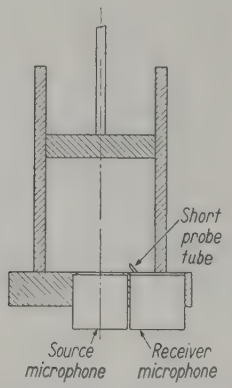


Fig. 5. Measuring apparatus with adjustable piston for control measurements.

The apparatus constructed were tested by measuring the impedance Z_{oc} of the closed tube as a function of the length at a fixed frequency.

Fig. 6 shows one curve measured at 2 kc/s. The source corresponds approximately to that shown in Fig. 2, i.e. $a_1/a = 0.4$, hence the position of the probe tube should be about $r/a = 0.6$ for an axial position $x/a = 0.1$. The full line is the computed impedance curve

$$\frac{Z_{oc}}{j\varrho c} = -\frac{\cos k\,(l-x)}{\sin kl},$$

and the agreement with the measured points is good, although the pressure varies about 15 dB at the measuring cross-section.

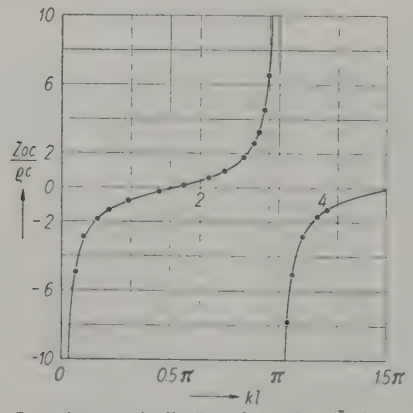


Fig. 6. Impedance ratio $Z_{oc}/j\varrho c$ of the closed tube, measured and computed for $x/a = 0.1$.

It would be desirable to reduce the measuring distance x/a as far as possible. On Fig. 7 is shown the computed impedance curve $Z_{oc}/j\varrho c = -\cot kl$ for $x = 0$ and the points shown are measured at 2 and 3 kc/s. It was found experimentally that the radial position of the probe tube in this case should be about $r/a = 0.55$, and this is in agreement with the computed pressure distribution curve for $x/a = 0$ on Fig. 2 shown in dotted line. The convergence of the formula is very slow for $x/a = 0$, so only part of the curve is given.

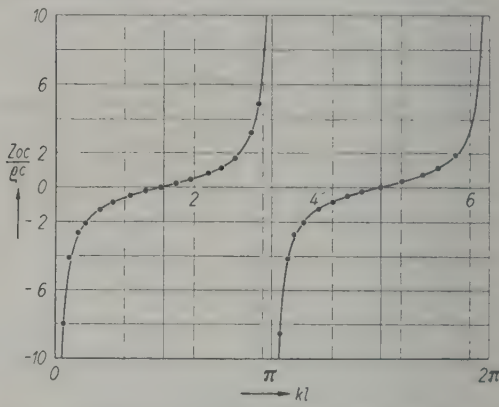


Fig. 7. Impedance ratio $Z_{oc}/j\varrho c$ of the closed tube, measured and computed for $x = 0$.

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[1] Beranek, L. L., Acoustic measurements. John Wiley & Sons, New York 1949, p. 313.
[2] Cook, R. K., J. acoust. Soc. Amer. 19 [1947], 922—923.
[3] Mawardi, O. K., J. acoust. Soc. Amer. 21 [1949], 84—91.
[4] Morse, P. M., Vibration and sound. McGraw-Hill Book Company, New York 1948, p. 307.
[5] Nielsen, A. K., Acoustic resonators. Trans. Danish Acad. tech. Sci. 10 [1949], 36.
[6] Nielsen, A. K., Acustica 2 [1952], 112—118.

Discussion

Morton, J. Y.: Mr. Nielsen's difficulty in obtaining a microphone with a good frequency response might be solved by using a moving coil probe tube microphone. At the British Post Office Research Laboratories we have moving coil instruments with a flat response up to 10 kc/s; Mr. West is reading a paper on these in this conference.

Lecturer: Such a microphone type has indeed been considered, but it was found difficult to obtain a high acoustic input impedance without reducing the sensitivity too much.

Morton, J. Y.: At the B.P.O. laboratory we have used a probe tube of internal diameter 0.9 mm and no difficulties have been experienced due to acoustic loading when measuring even small cavities.

Lecturer: The attenuation of such a probe tube is at least of the order 30 dB, and this is too much when a condenser microphone is used as a sound source, because the sound pressure produced is very limited.

MESURES DES COEFFICIENTS DE RÉFLEXION EN AMPLITUDE ET PHASE

par A. C. Raes
École Nationale Supérieure d'Architecture de Bruxelles

Sommaire. On décrit trois méthodes qui se sont révélées précieuses en acoustique architecturale. Leur principe essentiel est la séparation dans le temps et dans l'espace des ondes incidentes et réfléchies: 1) Essais avec ligne de transmission pour de petits échantillons sous incidence normale avec un appareil à lecture directe du coefficient de réflexion; 2) Essais en salle libre de matériaux absorbants, panneaux vibrants et rideaux sous incidence normale; 3) Essais sous incidence oblique. Extension de la méthode précédente.

Zusammenfassung. Es werden drei für die Bauakustik sehr nützliche Meßmethoden beschrieben, bei denen man im Prinzip die einfallende und reflektierte Welle zeitlich und räumlich trennt, so daß sie unabhängig voneinander gemessen werden können. So werden die Rechnungen auf ein Minimum beschränkt. Man kann bequem und mit großer Schnelligkeit kleine Proben bei senkrechtem Einfall prüfen. Ein direkt anzeigendes Meßgerät für den Reflexionskoeffizienten wird beschrieben. Weiter wird eine Konstruktion zur Messung schon befestigter Absorptionsmaterialien, einschließlich Plattenabsorber und Vorhängen, bei senkrechtem Einfall angegeben, wobei diese nicht berührt werden; dazu als Weiterentwicklung eine Methode für Tests bei schrägem Einfall.

Summary. A description is given of three methods which proved to be useful in architectural acoustics. The main principle is to separate the incident and the reflected waves in time and space so that they can be measured independently of each other. Calculations are thus reduced to a minimum. 1) Acoustical transmission line tests. Convenient to test small samples under normal incidence. Allows great operational speed. A direct reading reflection coefficient meter is described. 2) Free room tests. Designed to test absorbing materials, including vibrating panels and curtains, in situ and without touching to them, under normal incidence. 3) Oblique incidence tests. Development of the preceding case.

1. But

Les méthodes de mesures décrites ci-après ont pour but essentiel la mise au point et le contrôle des matériaux et des travaux. Nous avons donc cherché la rapidité d'opération avant la précision. Ces méthodes ont encore été élaborées en vue d'examiner le comportement de certains matériaux absorbants en régime transitoire.

2. Caractéristiques des impulsions utilisées

Nous avons adopté comme standard des impulsions constituées par une note pure modulée suivant une enveloppante composée de deux exponentielles. L'amplitude de l'onde de pression, pour la période croissante de l'impulsion a pour expression

$$P_c = P_m\,(1 - e^{-t/\tau}).$$

Pour la partie décroissante cette enveloppante s'exprime par

$$P_d = P_m\, e^{-t/\tau},$$

P_m étant l'amplitude maxima, t étant le temps et τ une constante. La figure 1 est l'enregistrement d'une telle impulsion. Nous appelons durée d'une impulsion le temps compris entre l'instant où P_c part de 0 et celui où P_d est revenu à 0,02 P_m (— 34 dB). Les durées pratiquement utilisables varient de 10 à 30 ms.

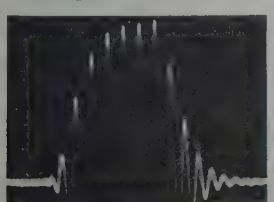


Fig. 1. La forme d'impulsion utilisée.

Ces formes d'impulsions ont les avantages suivants. a) Elles ont une expression mathématique exacte. b) On peut les produire avec un appareillage simple et robuste.

3. Mesures en incidence normale sur petites éprouvettes

a) Principe

Séparer dans l'espace et le temps l'onde incidente de l'onde réfléchie. Mesurer individuellement leurs amplitudes P_+ et P_-. Nous pouvons ainsi déterminer le coefficient de réflexion g sans artifices mathématiques.

b) Appareillage

Un tube droit (Fig. 2), à surface intérieure aussi réfléchissante que possible, a une longueur $2l$ au moins égale à celle d'une impulsion (dans l'espace). Une extrémité est fermée par un haut-parleur L, l'autre par l'éprouvette E. Au centre est placé un microphone M raccordé à un oscilloscope.

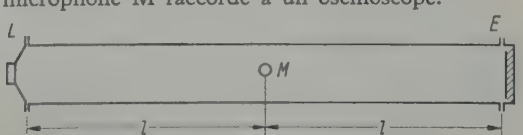


Fig. 2. Le dispositif à incidence normale.

c) Fonctionnement

L'impulsion émise par le haut-parleur se propage dans le tube et est d'abord enregistrée à son passage en M. Elle est ensuite réfléchie par l'éprouvette, et enregistrée à son second passage en M.

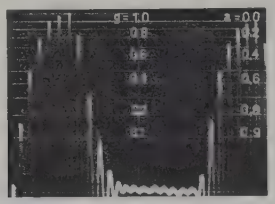


Fig. 3. Les impulsions incidentes et réfléchies par une plaque de verre.

Des réflexions se produisent encore successivement contre le haut-parleur et l'éprouvette. Nous ne les enregistrons pas. L'oscilloscope nous donne donc des figures analogues aux Fig. (3) et (4).

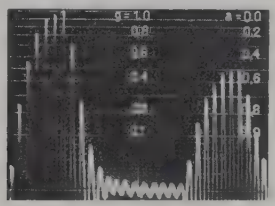


Fig. 4. Les impulsions incidentes et réfléchies par une couche absorbante.

d) Amplitude du coefficient de réflexion g

Si nous négligeons la constante d'atténuation par les parois du tube, nous pouvons admettre que les amplitudes A_+ et A_- des enregistrements des passages successifs d'impulsions sont proportionnels à P_+ et P_-, à la même échelle. Nous avons donc

$$\psi = -\frac{1}{2} \ln \frac{A_-}{A_+}$$

(L. L. Beranek, Acoustic measurements, p. 306).

Tenant compte de la constante d'atténuation, nous pouvons reprendre la théorie classique du tube, en tenant compte de ce que nous mesurons séparément P_+ et P_-.

e) Phase du coefficient de réflexion

Le déphasage des ondes réfléchies par rapport aux ondes incidentes peut être déterminé en mesurant les distances entre points correspondants de leurs enregistrements. La précision obtenue est de l'ordre du sixième de période, c'est à dire très faible.

f) Lecture directe des coefficients de réflexion et d'absorption

Tenant compte de ce que le coefficient d'atténuation du tube est très faible, nous avons admis que son influence état constante. Celà nous a permis de réaliser le dispositif suivant.

Devant l'écran de l'oscilloscope nous avons placé l'échelle visible aux figures 3 et 4. Les opérations deviennent extrèmement simples. Faire coincider l'axe des x de l'écran avec le trait inférieur de l'échelle, et ensuite régler l'amplification de manière à faire coincider le sommet de l'impulsion de gauche (A_+) avec le trait supérieur. Le sommet de l'impulsion de droite (A_) indique alors, sur l'échelle, le coefficient de réflexion g ou d'absorption a.

La figure 3 a été obtenue avec une plaque de verre épais. La figure 4 est le résultat de mesures sur un échantillon de panneau poreux. On lit $a = 0{,}51$.

g) Comparaison avec les résultats obtenus au tube de Kundt.

À titre d'exemple, citons que pour le panneau ayant donné la figure 4, nous avons obtenu 0,527 au tube de Kundt. Il est trop tôt pour généraliser définitivement.

h) Rapidité.

Le relevé des caractéristiques d'un échantillon, entre 200 et 3200 Hz, peut être fait en quelques minutes. Depuis que nous avons mis cette installation en service, nous avons pratiquement abandonné le tube de Kundt.

4. Mesures en incidence normale sur absorbants posés

a) Conditions à remplir par la méthode

Beaucoup de dispositifs absorbants, même des carreaux poreux perforés, agissent au moins partiellement par déformation élastique.

De plus dans de nombreux cas, par exemple pour des panneaux perforés devant des lames d'air, il faut réduire l'influence de la périphérie de la région observée. Celà s'obtient en augmentant la superficie de cette dernière.

Pour être d'utilité pratique générale, notre dispositif doit pouvoir agir sur un mètre carré au moins.

On ne peut pas appliquer un appareil contre le revêtement observé. Il faut éviter de l'abîmer. De plus, on doit pouvoir le laisser vibrer librement. Dans le cas de tentures ou de capitonnages, prendre appui est impossible.

b) Principe

Dans le dispositif précédent, donnons aux parois du tube un coefficient d'absorption égal à 1,0 à toutes les fréquences.

c) Appareillage

Sur une même perpendiculaire à la paroi examinée (Fig. 5), on place un microphone M et un haut-parleur émetteur d'impulsions L. La distance l_1 doit encore être suffisante pour que l'impulsion incidente et sa réflexion soient enregistrées séparément. Il est préférable de prendre $l_2 = l_1$. D'autre part, les autres parois du local doivent être suffisamment

distantés de L et M pour que leurs réflexions atteignent ce microphone sans se superposer à celle du mur ou plafond examiné. Le local doit donc avoir des dimensions minima d'environ 8 m dans tous les sens. Cette condition est toujours réalisée. On peut d'ailleurs encore réduire les dimensions parallèles au mur essayé en prenant des appareils directionnels. Il est d'ailleurs nécessaire de prendre un microphone directionnel orienté vers le mur. De cette façon l'enregistrement de l'onde incidente n'a pas une amplitude exagérée par rapport à celui de la réflexion.

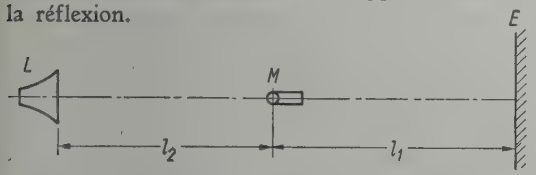


Fig. 5. La méthode des impulsions en champs libre.

d) Étalonnage

Une première méthode consiste à conserver toujours rigoureusement les mêmes positions relatives des appareils et des parois, ce qui ne demande qu'un peu de soin. On étalonne l'appareillage en laboratoire. Devant un mur très réfléchissant, on mesure une fois pour toutes le rapport A_-/A_+. Les valeurs de ce rapport pour des parois absorbantes sont facilement calculées. On trace la courbe g = fonction de A_-/A_+. Une seconde méthode consiste à emporter sur place un panneau réflecteur étalon que l'on place devant le mur. Les dimensions de ce panneau sont de l'ordre du mètre. Il reste maniable.

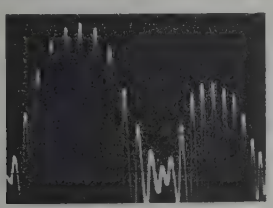


Fig. 6. Impulsion en champs libre et réflexion par un mur peint.

e) Exemples

La figure 6 donne un enregistrement de réflexion par un mur plafonné et recouvert de peinture émail. La figure 7 est obtenue en plaçant, devant ce même mur, un panneau de 0,6 × 6 m² du produit utilisé pour obtenir la figure 4. On trouve $a = 0{,}47$.

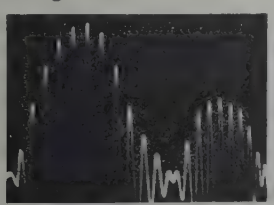


Fig. 7. Impulsion en champs libre et réflexion par un absorbant.

f) Utilité pratique

Citons un exemple. À l'aide de cette méthode, nous avons pu rapidement déceler qu'un enduit poreux avait été taloché une fois de trop, et obtenir des aveux de l'entrepreneur.

5. Mesures en incidence oblique sur absorbants posés

a) Principe

La méthode ci-dessus convient également à la mesure des coefficients de réflexion en incidence oblique. Deux dispositions sont appliquables. Lorsque l'incidence n'est pas trop écartée de la normale, on emploiera avantageusement la disposition schématisée à la figure 8.

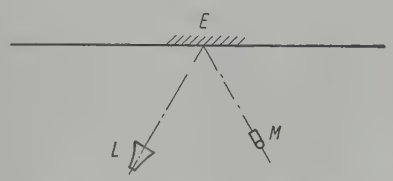


Fig. 8. La mesure de l'absorption sous incidence oblique.

Il faut que la longueur totale LEM dépasse au minimum de 4 m la distance en ligne droite LM.

En cas d'incidence très oblique, nous appliquerons le dispositif de la figure 9. Nous ajoutons un réflecteur auxiliaire R qui peut avoir des dimensions restreintes.

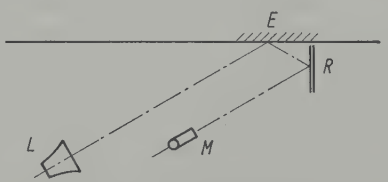


Fig. 9. La mesure de l'absorption à grandes angles d'incidence.

b) Appareillage

On peut utiliser le même appareillage que pour la méthode précédente. Le réflecteur R peut être un simple panneau de contreplaqué verni convenablement entretoisé.

Discussion

MAWARDI, O. K.: Doesn't the shape of the reflected pulse get to be so distorted that the evaluation of the maximum value becomes difficult?

CONFÉRENCIER: No; the reflected pulse does not become appreciably distorted, at least not when the reflecting element is a resistance (sample of rigid porous material), or an inductance and capacity (rigid wall with resonator).

BORDONE, C.: A method is developed to analyse the acoustical behaviour of small studios by the same impact sound system exposed by Mr. RAES, and to resume in a single representation the results of the analysis at varying frequencies or at varying directions of the sound source.

OPTICAL METHODS OF MEASUREMENT IN MOVING AIR STREAMS

by E. J. RICHARDS
University of Southampton, England

Summary. This paper describes a technique used in the Aeronautics Department of the University of Southampton in research on noise from jets. A new Schlieren system has been developed which allows the measurement of the velocities and frequencies of eddies moving downstream which are situated on the edges of both two and three dimensional jets. A focussing Schlieren method is put forward in conjunction with this technique in which the turbulence or eddying flow at a discrete point inside the jet may be measured. Applications of the technique to shock wave oscillations, speed measurement etc. are suggested.

Sommaire. On décrit dans cet article une méthode employée au Département Aéronautique de l'Université de Southampton pour étudier le bruit produit par des jets. On a mis au point un nouveau système de striation qui permet de mesurer la vitesse et la fréquence des tourbillons se déplaçant vers l'aval, et situés aux bords des jets de deux ou trois dimensions. On présente, en même temps que cette méthode, un procédé de focalisation des stries permettant de mesurer la turbulence ou l'écoulement tourbillonnaire en un point défini de l'intérieur du jet. On propose d'appliquer cette technique aux oscillations de l'onde de choc, à la mesure de la vitesse, etc.

Zusammenfassung. Es wird eine Meßmethode beschrieben, die in der Luftfahrtabteilung der Universität Southampton zur Untersuchung von Strahlgeräuschen verwendet wurde. Ein neues Schlierensystem wurde entwickelt, mit dem man die Geschwindigkeiten und Frequenzen von mitlaufenden Wirbeln an den Rändern sowohl zwei- als auch dreidimensionaler Strahlen messen kann. Im Zusammenhang mit dieser Versuchstechnik wurde eine fokussierende Schlierenmethode gefunden, mit deren Hilfe die Turbulenz oder der Wirbelfluß an einem bestimmten Punkt innerhalb des Strahles bestimmt werden kann. Es werden Anwendungen dieser Technik für Stoßwellenschwingungen, Geschwindigkeitsmessungen usw. vorgeschlagen.

1. Introduction

Noise research at the University of Southampton is aimed at furthering our understanding of the formation of noise from a jet stream issuing from a nozzle at supersonic speeds. In this condition, we have found that the noise spectrum is not completely random but carries, under various conditions, discrete notes which can be related to the stream conditions. Indeed, POWELL has formulated a most interesting "back reaction" theory in which the frequencies of the discrete notes can be predicted. According to this theory, any periodic disturbance, travelling downstream, gives rise to a sound wave on passing through a standing shock, which in turn travels upstream and on reaching the nozzle exit causes sufficient pressure change to initiate the next disturbance. Thus a resonance mechanism is set up which gives a sound field of discrete notes and a directional spectrum which is reasonably in accord with practical measurements.

In the establishment of this mechanism of noise production in both two dimensional and circular choked jets, it has been necessary to establish a new technique to measure the speed and frequency of the eddies moving downstream since the relative small size of one of the jets used (3 mm) has brought with it eddy frequencies above those measurable with existing hot wire techniques, jet velocities above and below that of sound, and sound frequencies somewhat greater than those that can be accomodated on a normal noise meter. For this latter purpose a new noise analyser has been developed which will be described elsewhere: the purpose of this paper is to outline the new techniques used and about to be used to measure the disturbance frequencies and velocities on the edge of both two and three dimensional jets.

2. Technique

The optical technique initially used was essentially that of a standard Schlieren system, the image being projected on a screen in the usual way. In this screen however was located a small pin-hole at the point on the image corresponding to the region in the stream where the analysis of the fluctuations were required. The varying light thus penetrating the aperture was picked up by a photomultiplier unit,

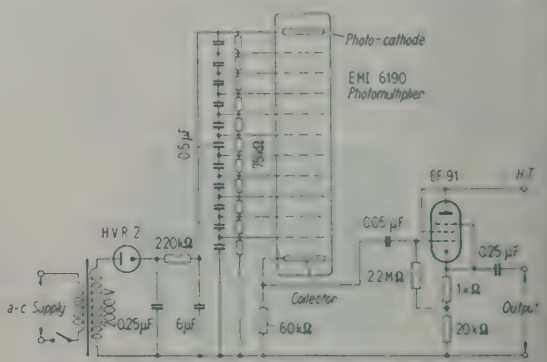


Fig. 1. Circuit diagram for photo-electric device.

the circuit system of which is shown in Fig. 1, and the output of which was displayed on a cathode ray oscillograph. Thus any dominating frequencies could be observed instantaneously by photography or they could be compared with the corresponding frequency noted with a microphone by comparative display.

By a simple extension of this technique, a second pinhole downstream of the other was used. The wavelength of the disturbances could thus be deduced from the magnitude of the signal, and it was a simple matter to assess the speed of travel downstream of the eddy or region of turbulence.

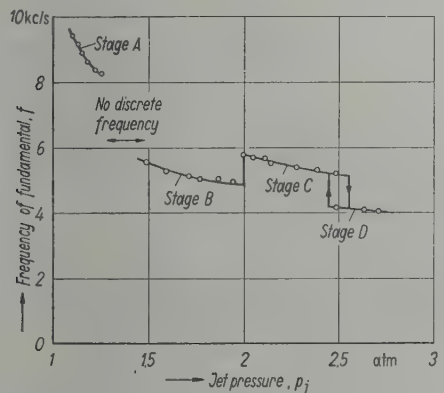


Fig. 2. Frequency of stream disturbance in axially symmetric case. Jet diameter = 25.4 mm.

As an illustration of the value of such a technique, Fig. 2 shows a plot of the frequency of the fundamental stream disturbance in the axially symmetric case at the edge of a circular 25.4 mm jet. In Fig. 3 the corresponding relationship of the wavelength-cell length ratio is plotted against square root of the excess pressure. In this case, the inter-relationship of the fundamental sound wavelength and the distance apart of the standing shock waves is clearly demonstrated.

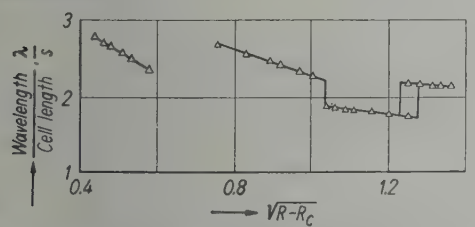


Fig. 3. Relation between fundamental radiated wavelength and cell length in axially symmetric case.
R = pressure ratio,
R_c = critical pressure ratio.

3. Advantages

The technique described here so briefly has many advantages over other optical and non-optical techniques of measuring eddy velocity and frequency.

a) The system deals most easily with the high frequencies obtained, and so allows the use of smaller scale jets in which the dominant frequencies would otherwise be too high to cope with.

b) The system is infinitely simpler than that of using a high speed camera which would need considerable development at the desired frequencies. No suitable camera is available in the United Kingdom.

c) Hot wire techniques cannot cope with the frequencies required and in any case may interfere with the air flow at the point under consideration. Recent reports from the U.S.A. however indicate considerable development in hot wire technique since this work began.

The great disadvantage so far is the non-focussing nature of the technique although this is not serious if it is the large scale turbulence at the edge of the jet which is being examined. A focussing system is being considered and is shown diagrammatically in Fig. 4. The parallel light system is focussed to pass

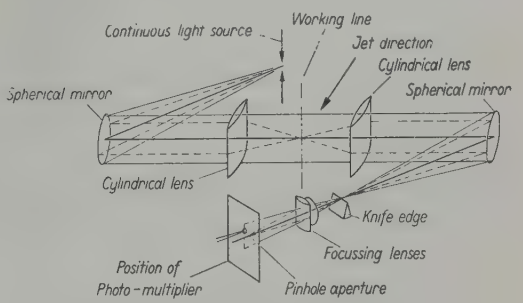


Fig. 4. Diagram of the focussing system.

through a vertical line in the jet by the use of two cylindrical lenses, the light then proceeding in the usual way via a mirror to a focus. The cut-off is arranged so that light directed in the vertical plane is let through and the Schlieren picture is reduced to a vertical line again by a third cylindrical lens. The fluctuating light at any predetermined point on this line is measured on the photomultiplier unit and is suitably amplified and displayed on the oscillograph screen. Since in this way all the deviated light is focussed to a line and all the light goes through the focussing line in the jet, it is argued that the variation measured on the oscillograph, will be predominantly related to the variation at the corresponding point, the light deviation arising from other areas of the jet cancelling out in the mean.

4. Other applications

a) The original system has sufficient sensitivity to measure not only the fluctuations in turbulence but also the actual sound waves as photographed with the Schlieren technique. This optical method

of measuring noise, needless to say, eliminates the problem of microphone response at high frequencies and may well have many applications in the high frequency region, but the form of the acoustic field must be known and calculable.

b) The system can be applied (without the introduction of anything in the stream) to the problem of shockwave oscillations over an aerofoil or in a wind tunnel.

c) If variations of density are put into the stream (e.g. by using sparks), it should be possible to measure local velocity in a supersonic wind tunnel far more easily than with existing techniques.

All these applications are to be looked at when time permits at the University of Southampton.

Discussion

PAHUD, J. PH.: a) Quelle est la profondeur de champ du strioscope focalisant décrit? b) La méthode de focalisation par lentilles cylindriques décrite ici est utilisée dans un strioscope à deux miroirs. Pourrait-elle s'appliquer à un strioscope à un miroir construit selon le principe dit "de coïncidence"? Cela serait intéressant, car le strioscope à un miroir est beaucoup plus sensible du fait que la lumière traverse deux fois la perturbation étudiée.

LECTURER: a) La profondeur de champ n'a pas été étudiée. b) Il semble que cela soit applicable, mais on perdrait en netteté. Je n'ai pas connaissance que cet essai ait été tenté.

GAVREAU, V.: L'effet des perturbations peut se manifester tout le long du trajet de la lumière entre les deux lentilles cylindriques. Pourquoi croyez-vous que seules les perturbations dans la région du foyer seraient détectées par ce procédé?

LECTURER: Les perturbations provenant des régions autres que le foyer sont distribuées au hasard (random distribution), et pour cette raison elles se compensent mutuellement. Ainsi seules les perturbations au foyer des lentilles cylindriques sont détectées.

ZWIKKER, C.: You explained the occurrence of sound because of sonic signals travelling upstream from the first shock wave to the nozzle. Is that possible as the velocity of the jet is supersonic?

LECTURER: The sound wave travels upstream in the nearly static air surrounding the main jet stream. The velocity of propagation will therefore be approximately that of sound.

RADEMAKERS, A.: Referring to the question of M. PAHUD it seems to me possible to put a plane mirror behind the second cylindrical lens, sending the light back along the same path. Separation will then be possible by a half-reflecting mirror, and thus the focal line can be the same for both incoming and returning light.

LECTURER: As a general rule sufficient sensitivity is provided by allowing the light through the working point once only. There are of course many Schlieren systems that use the principle mentioned by the questioner, but in every case the clarity of the image is affected adversely.

ABSOLUTE CALIBRATION OF HYDROPHONES BY VARIOUS METHODS IN THE FREQUENCY RANGE 0.1 ··· 15 KC/S

by K. TAMM

assisted by TH. LANGE and J. MALZFELDT

III. Physikalisches Institut der Universität Göttingen

Summary. Three types of piezoelectric pressure-sensitive hydrophones suitable for calibration have been described, a diaphragm type (ADP-crystals), an oil capsule type (LS-crystals), and a tourmaline receiver. The calibration methods: pistonphone (130 c/s), reciprocity cavity (0.2 ··· 3 kc/s), field calibration by RAYLEIGH disc (10 ··· 15 kc/s) result in a frequency-independent sensitivity of the oil capsule type of 10^{-4} V/(N/m²). An electrostatic relative calibration (1 ··· 20 kc/s) completes the frequency range.

Sommaire. On décrit 3 types d'hydrophones piézoélectriques étalonnables, pour mesure de la pression: un type à membrane (cristaux ADP), un type à capsule d'huile (cristaux LS), et un récepteur à tourmaline. L'emploi de différentes méthodes d'étalonnage: pistonphone (130 Hz), cavité à réciprocité (de 0,2 à 3 kHz), étalonnage en champ par le disque de RAYLEIGH (de 10 à 15 kHz) montre que le type à capsule d'huile a une sensibilité indépendante de la fréquence et valant 10^{-4} V/(N/m²). L'intervalle restant dans la gamme des fréquences est étalonné d'une façon relative par la méthode électrostatique (de 1 à 20 kHz).

Zusammenfassung. Drei Arten von eichbaren piezoelektrischen Schalldruck-Empfängern für Wasserschall werden beschrieben, ein Membranmodell (ADP-Kristalle), ein Ölkapselmodell (LS-Kristalle) und ein ungekapselter Turmalinempfänger. Die verschiedenen Eichverfahren: Pistonphon (130 Hz), Reziprozitäts-Druckkammer (0,2 ··· 3 kHz) und Feldeichung mittels RAYLEIGHscheibe (10 ··· 15 kHz) ergeben übereinstimmend eine frequenzunabhängige Empfindlichkeit des Ölkapselmodells von 10^{-4} V/(N/m²). Die Frequenzlücke wird durch eine elektrostatische Relativeichung (1 ··· 20 kHz) überbrückt.

Absolute calibration of sound receivers means coordination of the original sound field value (e.g. sound pressure) to the measured electrical output (e.g. voltage). It is given by a single figure (the sensitivity) in the case of a linear relation. Only receivers which are invariable with time and do not depend

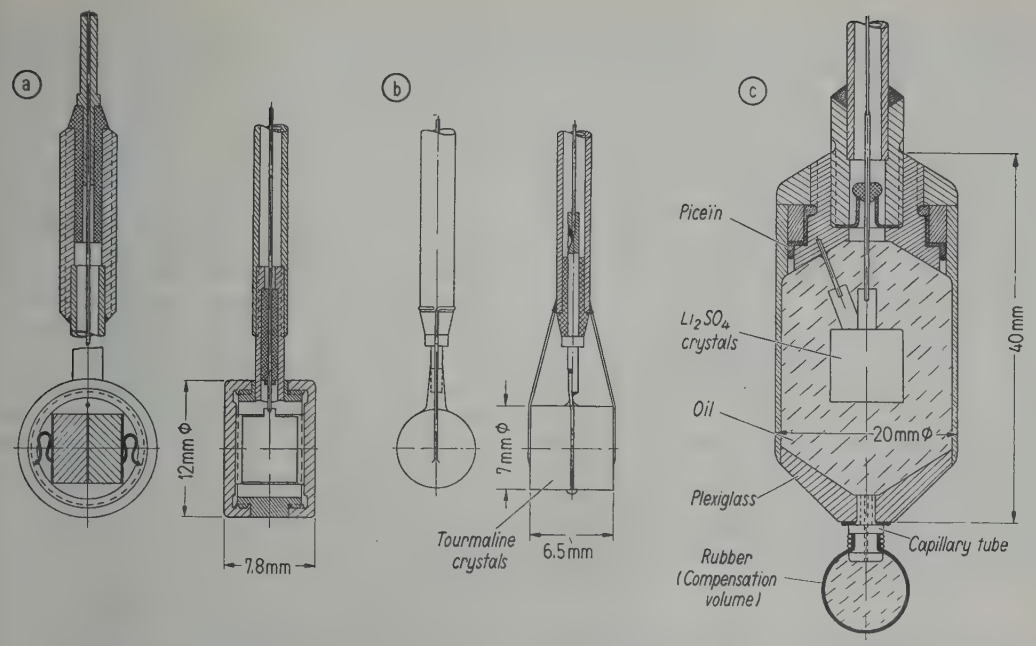


Fig. 1. Design of three different types of hydrophone for measurement purposes: (a) diaphragm type (ADP), (b) tourmaline or barium titanate type, (c) oil capsule type (LS).

too much on temperature and frequency are suitable for calibration. They should disturb the sound field as little as possible i.e. they should be small compared with the wavelength and incompressible compared with the medium.

According to these principles three types of piezoelectric pressure-receivers have been developed (Fig. 1): a diaphragm type with ADP-crystals, an oil capsule type with lithium sulphate crystals and a receiver without any box with tourmaline crystals or with barium titanate. To avoid low resonance frequencies no bending systems were used but only "thickness vibrators". The sensitivity is therefore low and amounts to 10^{-4} V/(N/m²) for the first two types, a value which is 100 times lower than that of a normal condenser microphone.

The *diaphragm type* consists of a cylindrical box and two screw caps between which two ADP-crystals are clamped. The sound pressure therefore acts upon the crystals only in one direction as is necessary for a piezoelectric output with ADP-crystals.

Tourmaline crystals and polarized barium titanate can be used without any box because of their hydrostatic piezoelectric effect. For insulation a film of rubber or lacquer is sufficient. Such receivers can be made very small (a few millimeters in diameter and height) but their sensitivity is too low for calibration by the methods described.

Lithium sulphate crystals give a higher "hydrostatic" effect but because of their solubility in water have to be used within a box which is filled with a non-aggressive liquid (castor oil) transmitting the sound pressure to the crystals. The box will be acoustically ineffective if it is made from plastic, rubber, or similar materials of the same characteristic impedance as water. The box of the hydrophone used for calibration was made of plexiglass and had the form of a cylinder with conical ends. It was metallized on its outside for electrical screening and to make it suitable for an electrostatic calibration. Because of its good acoustical qualities, especially this type of receiver was used for calibration.

An absolute calibration over a wide frequency range cannot be made by one method only. Therefore the calibrations were made by different methods each used at a single frequency or in the frequency range it was suitable for. For low frequencies the receiver is small compared with the wavelength and does not disturb the sound field because of its incompressibility. Therefore the sound pressure is constant over the whole surface of the receiver and the electrical output, being independent of the form of the sound field, is governed by the sound pressure which has to be calculated (pressure calibration). Hydrophones therefore can be calibrated in airfilled cavities which however have to be large enough to enclose the whole receiver and not only a diaphragm as is sufficient for airborne condenser microphones. This condition therefore limits the frequency range of the pressure calibration for the oil capsule type

receiver below 3 kc/s. For higher frequencies the form of the sound field is of influence and a field calibration has to be carried out. Because of the low sensitivity of the hydrophones, relatively high sound pressures have to be used for calibration.

At the lower frequency limit the calibration was made by means of a pistonphone. The large sinusoidal amplitudes of the piston necessary for a sufficiently high sound pressure were produced by using the resonance (130 c/s) of the vibrating piston with its suspension spider.

For adiabatic compression (ratio of the specific heats $\varkappa = 1.402$) the alternating pressure p in the cavity can be determined from the optically measured amplitide $a\sqrt{2}$ of the piston as a percentage of the atmospheric pressure P_0 if the volume of the cavity V_0 (= 101.4 cm³) and of the receiver V_R and the cross-section S (= 0.1248 cm²) of the piston are known. The inaccuracy will not exceed 1.4%, if a correction ($\varepsilon = 0.6\%$) is made for the heat conduction of the walls.

$$p = P_0 \cdot (1 - \varepsilon) \cdot S \cdot a/(V_0 - V_R).$$

In the frequency range 0.2 ··· 3 kc/s a reciprocity calibration in a cylindrical cavity was applied. Two electrostatic systems with solid dielectric (SELL systems) forming the plane circular walls of the cavity are used for auxiliary source and reversible transducer. The receiver to be calibrated is put in the middle between the two systems (Fig. 2). In

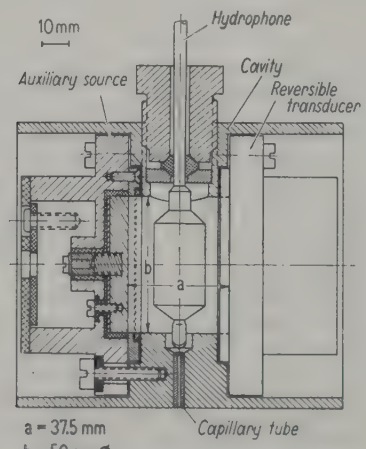


Fig. 2. Test cavity for reciprocity calibration of small microphones and hydrophones with electrostatic systems for auxiliary source and reversible transducer.

the characteristic method for the reciprocity calibration, the sensitivity E_{0R} of the receiver is determined by comparing the open circuit voltages e_0 of the actual receiving system in the three combinations (transducer — receiver: e_0', auxiliary source — receiver: e_0'', auxiliary source — transducer: e_{0T}) with the voltage iR caused by the (constant) current through the actual transmitting system at the known resistance R. If the form of the sound field is taken into account by the reciprocity factor $|J_0 K_1 K^2|$ there results:

$$E_{0R} = \left|\frac{e_0'}{Ri_T}\right| \cdot \left|\frac{e_0''}{Ri_S''}\right| \cdot \left|\frac{Ri_S'''}{e_{0T}}\right| \cdot R \cdot |J_0 K_1 K^2|.$$

In this formula $|J_0| = A/\varrho c \cdot \sin(H\omega/c) \approx \omega AH/\varrho c^2$ means the reciprocity factor between auxiliary source and transducer, if there is no reaction between the systems and the sound field. It corresponds to the reciprocal of the stiffness reactance of the air column with the height H and the area A at the pulsatance ω. If the volume V_R of the receiver is not negligible, $AH - V_R = V'$ is to be put in place of AH. The factor $K_1 = \sin(H'\omega/c)/\sin(H\omega/c)$ takes regard of the mutual reaction between systems and air volume by a correction $\Delta H = H' - H$ in height, which however in the apparatus is only 0.01 mm. The correction factor $K = 1/\cos(H'\omega/2c)$ is caused by the fact that the receiver is placed at the centre of the cavity. For frequencies higher than 1 kc/s all corrections are to be considered (results s. Fig. 4):

$$J_0 K_1 K^2 = \frac{\omega V'}{\varrho c^2} \cdot \frac{\tan(H'\omega/2c)}{H'\omega/2c}.$$

In the high frequency range 10 ··· 15 kc/s a field calibration in a flat horizontal layer of water (flat basin [1]) was applied. For a depth H of the water, only a little larger than a half of the wavelength λ, the attenuation of the sound with distance is large enough to make reflections at the wall ineffective, therefore a damped progressive cylindrical wave is produced by a point source. Its phase velocity $c' = c/\sqrt{1 - (\lambda/2H)^2}$ is higher than that in the free medium and can be easily measured. It determines the ratio of sound pressure to horizontal particle velocity v, i.e. the characteristic impedance of the layer $\varrho c'$. This value enables us to calculate the sound pressure from a measurement of the horizontal component of the particle velocity by means of a RAYLEIGH disc at that place to which the receiver will be brought afterwards (results s. Fig. 4).

Because of the finite ratio of densities of water (ϱ) and the material of the disc (ϱ_D) the influence of the mass (m_W) of the displaced water and the hydrodynamic mass $m' = 4\varrho a^3/3$ has to be taken into account by WOOD's correction of KOENIG's formula [2], [3], and for 45° incidence the torsional moment Da is given by:

$$Da = \varrho v^2 \cdot \frac{4}{3} a^3 \cdot \left(\frac{m_D - m_W}{m_D + m'}\right)^2.$$

For the ratio thickness (s) to radius (a) of the disc a compromise has to be chosen, because on one side the above formula holds only for thin discs and on the other side a minimum ratio ($s/a = 1/20$

for brass with $\varrho_D = 8.5\,\varrho$) is necessary to avoid flexural vibrations in the frequency range under test and to get the hydrodynamic mass smaller than the mass of the disc. The suspension of the disc finally used ($2a = 1.6$ cm, $s = 0.277$ cm) was chosen with respect to a suitable period of vibration i.e. 17.6 s.

The frequency range 3 ··· 10 kc/s still vacant between the reciprocity and RAYLEIGH disc calibration could be bridged by a relative calibration using the electrostatic method. For this purpose a constant a.c. voltage and a constant d.c. voltage are applied between the metallized surface of the box of the receiver and a uniformly spaced auxiliary electrode surrounding it (see Fig. 3). The a.c. forces to the surface cause an a.c. (sound) pressure inside the box which does not depend on frequency (results s. Fig. 4). By an exact determination of the air gap, an absolute calibration may be obtained also from this method.

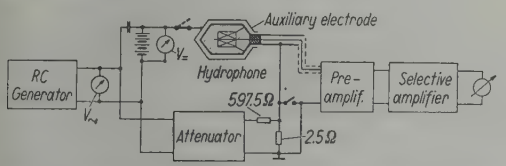


Fig. 3. Scheme of the electrostatic method used for (relative) calibration of the oil capsule hydrophone.

From Fig. 4 it may be seen that the results of the different methods of calibration agree very well. In the frequency range in question the sensitivity of the hydrophone (with LS crystals) under test was found independent of frequency to be — 99 dB (re. to 10 V/(N/m²) i.e. re to 1 V/μbar with an accuracy of ±0.5 ··· 1 dB.

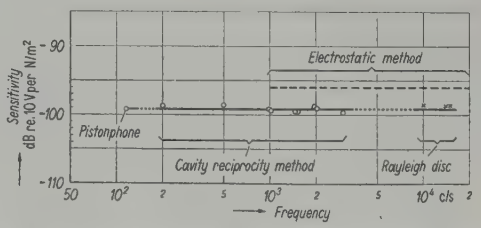


Fig. 4. Frequency response of the sensitivity of an oil capsule hydrophone as a result of absolute calibrations by different methods.

The work described in this report was carried out under arrangements with the Department of Scientific and Industrial Research, London, whose assistance and interest are gratefully acknowledged. It is published with the permission of the Departement.

References:

[1] TAMM, K., Akust. Z. 6 [1941], 16.
[2] WOOD, A. B., Proc. phys. Soc. 47, 5 [1935], 779.
[3] KOESTERS, A., Acustica 2 [1952], Beiheft 3, AB 171.

Discussion

GAVREAU, V.: Comment absorbez-vous le son à l'extrémité éloignée de la cuve? Nous utilisons un canal ouvert et une plage élargie, arrondie à son extrémité.

LECTURER: Une partie de l'énergie sonore est absorbée par les parois de la cuve et le reste par une structure absorbante à pyramides multiples à l'extrémité éloignée de la cuve.

PROBE MICROPHONES AS LABORATORY STANDARDS

by W. WEST

British Post Office, Engineering Department, London

Summary. To minimize standing waves in the probe tube, the "large-bore" probe microphone detects sound pressures on the wall of the tube (at a point remote from the open end) and the tube is terminated to eliminate reflexions. The tube of the "small-bore" probe microphone terminates at a small enclosure (the coupler) such that the acoustical impedances of tube and coupler are approximately equal.

Sommaire. Pour réduire les ondes stationnaires du tube sonde, dans le cas du microphone sonde à "grand alésage", on reçoit les pressions sonores sur la paroi du tube (en un point éloigné de l'extrémité ouverte), et l'extrémité de ce tube est conçue de façon à éviter des réflexions. Dans le cas du microphone sonde à "petit alésage", on termine le tube par une petite enceinte (de couplage) telle que les impédances acoustiques du tube et de cette enceinte soient sensiblement les mêmes.

Zusammenfassung. Um die stehenden Wellen im Sondenrohr möglichst klein zu halten, wird beim Sondenmikrophon mit weitem Rohr der Schalldruck an der Rohrwand (an einer vom offenen Ende entfernten Stelle) abgenommen und das Rohr reflexionsfrei abgeschlossen. Bei einem Sondenmikrophon mit engem Rohr endet das Rohr in einem kleinen Kopplungs-Hohlraum, dessen akustische Impedanz annähernd gleich der des Rohres ist.

About twenty years ago [1] the British Post Office became interested in a probe microphone for use as a laboratory standard to simplify measurements of free-field sensitivity. Such a microphone could be sufficiently small, at the point of application of sound, for a pressure calibration to be the same as

a free-field calibration. Thus the relatively tedious method of free-field calibration of microphones by direct comparison with a RAYLEIGH disk could be avoided, the pressure calibration of the probe microphone by RAYLEIGH disk being used instead.

The principle applied was to form the probe as a non-resonant tube by inserting sound absorbent at the far end, and to expose the sound pressures of waves passing along the tube to the centre part of the diaphragm of a condenser microphone. Fig. 1 illustrates the device as applied to a condenser microphone of the then conventional construction; the brass coupler fits closely over the diaphragm and has projections into which parts of a tube, 0.635 cm internal diameter, are fixed. One part, about 30 cm long, leads to the open end, or probe; the other part, about 150 cm long, has strands of wool drawn into the last 120 cm or so and is coiled on a 30 cm diameter. Within the coupler the tube is continued, altering shape but not cross-sectional area, to expose an oval aperture where the diaphragm acts as a part of the wall of the tube.

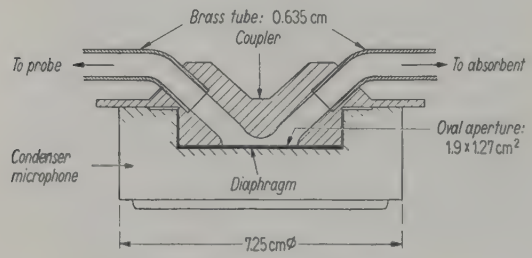


Fig. 1. Probe condenser microphone.

The sensitivity-frequency characteristic of this probe microphone is shown in Fig. 2. The distance along the tube to the centre of the coupler is about 33 cm, and the regular spacing (about 250 c/s) between peak and trough in the characteristic shows that there is some reflection from the coupler, resulting in the undulations, about $\pm$ 1.5 dB in extent. These do not lead to inaccuracies in measurements of a sound field, and the generally falling trend of the characteristic can easily be compensated in the microphone amplifier.

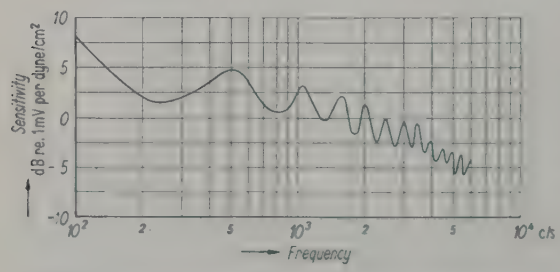


Fig. 2. Frequency characteristic of probe microphone.

A bore of 0.635 *cm* is too large for a probe used for measuring sound pressures where the acoustical impedance is large, e.g. in a small enclosure. ARCHBOLD [2] has developed the design of a probe microphone having a tube of only 0.77 mm internal diameter coupled to the dome-shaped diaphragm of a moving-coil microphone, the general construction being as shown in Fig. 3. His early experiments showed that, with a tube length of 12.7 cm, the frequency spacing between the peaks and troughs of the frequency characteristic, was about 550 c/s. With the coupler then in use the undulations were about $\pm$ 5 dB in extent. These could be eliminated, at a sacrifice of sensitivity, by the introduction of a small amount of lamb's wool for absorbing sound at each end of the tube, but the sensitivity fell away sharply at frequencies around 5000 c/s.

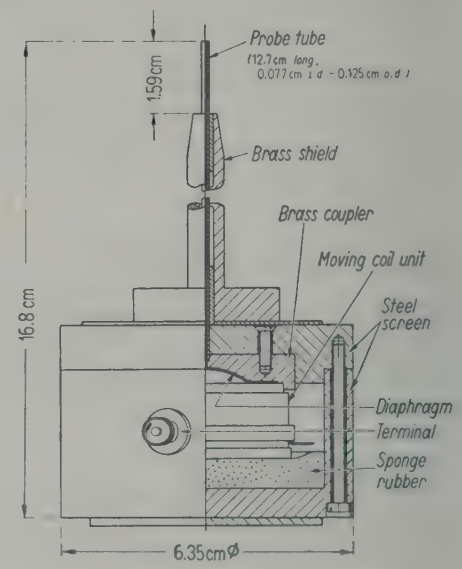


Fig. 3. Probe moving-coil microphone.

Further experiments were carried out with couplers having different curvatures of the domed surface and for different spacings between the coupler and the diaphragm at the edge and at the centre, separately. Reducing the spacing at the edge, up to a certain point, was found to increase the sensitivity at frequencies above about 3000 c/s. A radius of curvature of the coupler dome equal to that of the diaphragm appeared to be optimum. The group of curves in the upper part of Fig. 4 illustrates the effect, with these dimensions fixed, of varying the spacing between coupler and diaphragm at the centre. As this distance is decreased from 0.254 to 0.127 mm the undulations in the frequency characteristic decrease in extent. When it is further reduced to 0.102 mm not only do the undulations

increase again, but also the positions of peaks and troughs are interchanged with respect to the other curves. Evidently the impedance at the end of the tube passes through an approximate match to the characteristic impedance of the tube at a spacing of 0.127 mm. The lower part of Fig. 4 shows the requency characteristic of a similar microphone when the spacing is adjusted to optimum. No damping by lamb's wool is used.

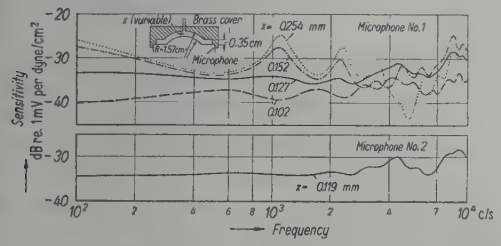


Fig. 4. Frequency characteristics of two probe microphones.

The sensitivities shown in Fig. 4 are measured at the output of a transformer of 1 to 50 turns ratio. The level of sensitivity is such as to enable sound pressures greater than about 1 dyne per cm² to be measured. Considerable increase of sensitivity can be obtained from each of three possible modifications. Moving-coil units of better sensitivity are now obtainable. A suitable transformer can be designed with higher turns ratio. The diameter of the tube can be increased somewhat; in this connexion it is of interest to quote some results obtained with a tube of 1.27 mm diameter. The centre spacing distance was varied in small steps over a range of 0.46 to 0.1 mm. Critical spacings (similar to the 0.127 mm position in Fig. 4) were found in three positions, viz. 0.33, 0.168 and 0.127 mm, and the interchange of position of peaks and troughs occured when passing through each. For the 0.127 mm position the lowest sensitivity at any frequency was — 30 dB.

References

[1] West, W., Post Office Elec. Eng. J. **26** [1934], 260.
[2] Archbold, R. B., ibid. **45** [1953], 145.

Discussion

Ingemansson, N. S. P.: One measurement with the tube damped showed a rather flat curve. Why was damping used in this case only?

Lecturer: Damping by lamb's wool was used in some early models; it is found to be unnecessary when the dimensions of the cavity are made correct for impedance matching.

Schenkel, G.: a) What are the acoustic resistors at the back of the diaphragm? b) Is the wave resistance of the tube the only element for damping?

Lecturer: a) All parts at the back of the diaphragm are as supplied in the commercial moving-coil unit. b) The small air-space in front of the diaphragm also adds damping.

Nielsen, A. K.: The temperature coefficient and the long-time stability of the probe microphone could be imagined to be inferior to the 640 AA type.

Lecturer: A maximum change of sensitivity of 0.3 dB has been measured due to a change of temperature of 10 °C for the probe microphone. The long-time stability at a given temperature is within 0.2 dB.

BEMERKUNGEN ZUR DEFINITION DES VEKTORIELLEN SCHALL-LEITWERTES, DES AKUSTISCHEN WIDERSTANDES UND DER AKUSTISCHEN ELEMENTE

von W. Willms
Forschungsinstitut Weil am Rhein, Deutschland

Zusammenfassung. Im Schallfeld können der Admittanzvektor und ein Impedanzvektor eingeführt werden. Hält man aber an der Reziprozitätsbeziehung $z = 1/y$ für die Komponenten der Vektoren fest, so kann man nur einen Vektor, vorzugsweise den Admittanzvektor, verwenden. Es wird ein quadratischer Flächenmittelwert für die Schnelle und den Druck im Querschnitt eines Wellenleiters vorgeschlagen. Während die direkte Analogie zwischen akustischen und elektrischen Widerstandselementen besondere Festsetzungen und Schaltvorschriften erforderlich macht, enthält die Kettenmatrix alle notwendigen Angaben und liefert die elektrischen Ersatzschaltbilder nach den beiden Analogiesystemen.

Summary. The admittance vector and an impedance vector may be introduced in the sound field. But, if for the vector components the relation of reciprocity $z = 1/y$ is to be maintained, only one vector, preferably the admittance vector, may be used. It is proposed to use in an acoustic waveguide a mean square surface value of particle velocity and pressure. The analogy between acoustic and electric impedance elements needs special definitions and rules. On the contrary the matrix gives all the information required and the two analogous circuits are easily derived.

Sommaire. On peut introduire dans l'expression d'un champ sonore le vecteur admittance et un vecteur impédance. Mais si on tient à conserver la relation de réciprocité $z = 1/y$ entre les composantes des vecteurs, on ne peut utiliser qu'un seul vecteur, de préférence le vecteur admittance. On préconise d'employer comme vitesse et pression dans la section d'un guide d'ondes la moyenne

quadratique prise sur la surface. Alors que l'analogie directe entre les éléments des résistances acoustique et électrique oblige à établir des définitions et des règles d'acoustique spéciales, la représentation en forme de matrices fournit toutes les données nécessaires et permet d'établir les schémas électriques équivalents dans les deux systèmes d'analogie.

Die Frage einer angemessenen Definition der spezifischen akustischen Impedanz ist neuerdings wieder aufgeworfen worden, weil die übliche sich nur auf den eindimensionalen Fall bezieht und in so enger Analogie zur elektrischen Impedanz eingeführt wird, daß man von einer eigentlichen akustischen Größe kaum sprechen kann. Im Fall des dreidimensionalen Schallfeldes und bedingt durch den verschiedenen Charakter des Schalldruckes als Skalar und der Schnelle als Vektor treten besondere Verhältnisse auf, die bei der Verallgemeinerung des Impedanz- und Admittanzbegriffes beachtet werden müssen. Dies ist der Gegenstand der ersten Bemerkung.

Bei der Begriffsbildung der akustischen Impedanz war die stationäre Rohrströmung sehr stark das Vorbild. Nun ist aber der quasistationäre Schallfluß durch einen Schalleiter nur bedingt als Strömung anzusehen. Vielmehr ist es die Energie, die im Schalleiter transportiert wird. Es erscheint demnach richtiger, an Stelle der üblichen linear gemittelten Flußgrößen quadratisch gemittelte einzuführen.

Die dritte Bemerkung steht nur in losem Zusammenhang mit den anderen. Bei der Beschäftigung mit akustischen Elementen empfindet man es oft als Nachteil, daß sie durch ihre „Impedanz" und durch ihre „Schaltung" gekennzeichnet werden, die je nach der verwendeten Analogie verschieden sind. Beide Eigenschaften der Elemente kann man in ihrer Kettenmatrix vereinen.

1. Impedanz und Admittanz im Schallfeld

Die spezifische akustische Impedanz in einem Punkte des Schallfeldes und in einer bestimmten Richtung wird als das Verhältnis des komplexen Schalldruckes p und der komplexen Komponente v der Schnelle in der betreffenden Richtung definiert:

$$z = \frac{p}{v}. \tag{1}$$

Das Reziproke hiervon ist die spezifische akustische Admittanz:

$$y = \frac{v}{p}. \tag{2}$$

Es besteht also die Beziehung

$$zy = 1. \tag{3}$$

Man kann nun, wie O. K. MAWARDI (J. acoust. Soc. Amer. 23 [1951], 571) dies getan hat, im dreidimensionalen Schallfeld den Admittanzvektor $\mathbf{y}$ einführen

$$\mathbf{y} = \frac{\mathbf{v}}{p}; \tag{4}$$

$\mathbf{y}$ und $\mathbf{v}$ sind komplexe Vektoren, p ist ein komplexer Skalar. Die übliche spezifische akustische Admittanz ergibt sich als die (komplexe) Komponente des Admittanzvektors in der betreffenden Richtung, ist also das skalare Produkt des Admittanzvektors $\mathbf{y}$ mit dem Richtungsvektor $\mathbf{n}$ (Richtungswinkel α, β, γ)

$$y = \mathbf{y} \cdot \mathbf{n} = \frac{1}{p}\,(v_x \cos\alpha + v_y \cos\beta + v_z \cos\gamma). \tag{5}$$

Es fragt sich nun, ob man auch die spezifische akustische Impedanz von einem ähnlichen Impedanzvektor herleiten kann.

Ein Impedanzvektor, der die Gl. (1) erfüllt, ist

$$p = \mathbf{z}^*\mathbf{v}. \tag{6}$$

$\mathbf{z}^*$ ist ein Zeilenvektor mit konjugiert komplexen Komponenten

$$\mathbf{z}^* = (\bar{z}_x\, \bar{z}_y\, \bar{z}_z). \tag{7}$$

Da p ein gegen Koordinatentransformationen invarianter Skalar ist und $\mathbf{v}$ aus kontravarianten Vektorkomponenten besteht, müssen die Komponenten von $\mathbf{z}^*$ kovariant sein, im Gegensatz zu $\mathbf{v}$, also einen Zeilenvektor aus konjugiert komplexen Komponenten bilden.

Der gesuchte Impedanzvektor, als Spaltenvektor geschrieben, lautet

$$\mathbf{z} = \begin{pmatrix} z_x \\ z_y \\ z_z \end{pmatrix}. \tag{8}$$

Mit Gl. (6) ist der $\mathbf{z}^*$-Vektor noch nicht genügend bestimmt, es gibt unendlich viele Vektoren, die Gl. (6) erfüllen. Fordert man wie MAWARDI weiterhin, daß der Impedanzvektor $\mathbf{z}$ ein Vielfaches λ des $\mathbf{v}$-Vektors sei, so ist der einzige Vektor, der Gl. (6) und die Zusatzforderung erfüllt,

$$\mathbf{z} = \frac{p}{|\mathbf{v}|^2}\,\mathbf{v}. \tag{9}$$

Denn mit

$$\mathbf{z} = \lambda\,\mathbf{v}, \quad \mathbf{z}^* = \lambda\,\mathbf{v}^*$$

folgt aus Gl. (6)

$$p = \lambda\,\mathbf{v}^*\mathbf{v}, \quad \bar{p} = \lambda\,|\mathbf{v}|^2$$

also

$$\lambda = \frac{p}{|\mathbf{v}|^2},$$

wie in Gl. (9) behauptet.

Dieser unter den genannten Voraussetzungen definierte Impedanzvektor hat den schwerwiegenden Nachteil, daß seine Komponente in einer gegebenen Richtung, also die spezifische akustische Impedanz, die Reziprozitätsbeziehung Gl. (3) nicht erfüllt. Es ist im allgemeinen

$$z = \frac{\bar{p}}{|v|^2}(v_x \cos\alpha + v_y \cos\beta + v_z \cos\gamma) \neq$$
$$\neq \frac{p}{v_x \cos\alpha + v_y \cos\beta + v_z \cos\gamma} = \frac{1}{y}. \quad (10)$$

Die Reziprozitätsbeziehung läßt sich nur für drei Richtungen, nicht aber in voller Allgemeinheit erfüllen.

Man muß sich entscheiden, entweder einen Admittanzvektor oder einen Impedanzvektor einzuführen. Aus Gründen der Einfachheit wird vorgeschlagen, nur den Admittanzvektor zu nehmen. Aus dem Admittanzvektor erhält man die spezifische akustische Admittanz in dem betreffenden Punkte und in der betreffenden Richtung des Feldes. Die spezifische akustische Impedanz ist dann als das Reziproke der spezifischen akustischen Admittanz zu definieren.

Im Schallfeld treten somit drei Vektorgrößen auf:
1. der Vektor der Schnelle $\mathbf{v}$
2. der Admittanzvektor $\mathbf{y} = \mathbf{v}/p$
3. der Poyntingsche Vektor $\mathbf{S}^* = \frac{1}{2}\bar{p}\,\mathbf{v} = \frac{1}{2}|p|^2\,\mathbf{y}$.

Alle drei Vektoren unterscheiden sich nicht wesentlich. Sie sind mit der Schnelle verwandt und nur in verschiedener Weise normiert.

2. Quadratischer Flächenmittelwert

Der Schallfluß q durch den Querschnitt eines Schalleiters wird im allgemeinen anschaulich als das Produkt der Schnelle v und der Fläche S eingeführt,

$$q = v\,S, \quad (1)$$

bei nicht konstanter Schnelle auch

$$q = \int v\,\mathrm{d}S. \quad (2)$$

Die akustische Impedanz des Schalleiters, z.B. seine Antriebsimpedanz, ist

$$Z = \frac{p}{q} = \frac{p}{vS}. \quad (3)$$

Mit dieser Definition kommt man in Schwierigkeiten, wenn Druck und Schnelle nicht mehr konstant über dem Querschnitt sind, wie z.B. bei Wellen höherer Ordnung. Hier ist es zweckmäßig, die akustische Impedanz energetisch zu definieren:

$$Z = \frac{\frac{1}{2}\int p\,\bar{v}\,\mathrm{d}S + 2\,\mathrm{j}\omega\,(T - V)}{\frac{1}{2}\,v_m^2\,S^2}$$
$$= \frac{\frac{1}{2}\,p_m^2}{\frac{1}{2}\int p\,\bar{v}\,\mathrm{d}S - 2\,\mathrm{j}\omega\,(T - V)}, \quad (4)$$

v_m und p_m sind noch näher zu bestimmende Mittelwerte der Schnelle und des Druckes, $\frac{1}{2}\int p\,\bar{v}\,\mathrm{d}S$ ist der Leistungsfluß durch das akustische System, $T - V$ der Unterschied zwischen der darin gespeicherten kinetischen und potentiellen Energie.

In einigen einfachen Systemen kann man die Leistung und die Impedanz angeben, so daß Gl. (4) dann als Bestimmungsgleichung für v_m und p_m angesehen werden kann. Ein Beispiel hierfür ist der unendlich lange Wellenleiter, der in einer Welle höherer Ordnung angeregt sein möge. Der Leistungsfluß ist reell, $T - V$ ist gleich Null. Andere Beispiele sind die kurze, mit $p = 0$ abgeschlossene Rohrleitung oder der mit $v = 0$ abgeschlossene Hohlraum, in denen überwiegend nur kinetische oder potentielle Energie gespeichert ist und in denen der Leistungsfluß Null ist. In allen Fällen wird man für v_m und p_m auf Ausdrücke von der Form

$$v_m = \sqrt{\frac{1}{S}\int |v|^2\,\mathrm{d}S}\,; \quad p_m = \sqrt{\frac{1}{S}\int |p|^2\,\mathrm{d}S} \quad (5)$$

geführt, die man in Anlehnung an den quadratischen Zeitmittelwert den quadratischen Flächenmittelwert der betreffenden Größe nennen kann.

Der quadratische Flächenmittelwert des Schallflusses $q_m = v_m S$ geht für konstante Schnelle bzw. für die Welle nullter Ordnung in den üblichen Wert $q = vS$ über.

Es erscheint angebracht, den quadratischen Flächenmittelwert auch bei der Berechnung des Schallflusses in engen Rohren zu verwenden, wo man bisher den arithmetischen Mittelwert nach Gl. (2) gebrauchte. Der Unterschied ist übrigens praktisch nicht groß. Z.B. ist bei parabolischer Verteilung der Schnelle der quadratische Flächenmittelwert nur um 9% größer als der arithmetische Mittelwert. Bei weiten Rohren ist der Unterschied noch kleiner.

3. Akustische Elemente

Unter akustischen Elementen sollen Rohrleitungsstücke verstanden werden, die von festen Wänden begrenzt und in allen Dimensionen klein zur Wellenlänge sind. Akustische Elemente bilden die Grundbestandteile der akustischen Filter.



Abb. 1.

Der Bewegungszustand in den Querschnittsverengungen ist quasistationär, in den Hohlräumen quasihomogen. Als komplexen akustischen Widerstand eines Elementes definiert man

$$Z = \frac{\Delta p}{q} \text{ (Rohrelement).} \quad (1)$$

Diese Definitionsgleichung ist nur für die Elemente mit quasistationärem Strömungszustand anwendbar. Für die mit quasihomogenem Zustand muß man eine neue Definition

$$Z = \frac{p}{\Delta q} \text{ (Hohlraumelement)} \quad (2)$$

einführen, denn bei diesen besteht ja kein Druck-

unterschied an den beiden Querschnitten, sondern nur ein Flußunterschied. Außerdem unterscheiden sich die beiden Widerstandselemente durch ihre Übertragungseigenschaften. Die Rohrleitung ist flußübertragend, die Hohlräume sind druckübertragend.

Elektrische Widerstandselemente wie Drosselspule und Kondensator haben dagegen nur eine Widerstandsdefinition

$$Z_{el} = \frac{\Delta\Phi}{I} = \frac{U}{I} \tag{3}$$

und sind beide stromübertragend. Die direkten Analogiebeziehungen zwischen akustischen und elektrischen Widerstandselementen sind daher erzwungen und erfordern bestimmte Voraussetzungen, Schaltvorschriften und Schaltverbote. Die Umwandlung akustischer Gebilde in elektrische Schaltungen ist möglich, dagegen nicht immer die Realisierung elektrischer Schaltungen durch akustische Elemente.

Viel müheloser ergeben sich die akustisch-elektrischen Analogiebeziehungen, wenn man die Übertragungsgleichungen anschreibt

$$\begin{aligned} p_1 &= p_2 \\ q_1 &= Z^{-1}\, p_2 + q_2 \end{aligned} \quad \text{(Hohlraum)} \qquad \begin{aligned} p_1 &= p_2 + Z q_2 \\ q_1 &= q_2 \end{aligned} \quad \text{(Rohr)} \tag{4}$$

oder als eine Art Kurzschrift das Koeffizientenschema, die Kettenmatrix

$$\left\| \begin{matrix} 1 & 0 \\ Z^{-1} & 1 \end{matrix} \right\| \text{(Hohlraum)} \qquad \left\| \begin{matrix} 1 & Z \\ 0 & 1 \end{matrix} \right\| \text{(Rohr)} \tag{5}$$

In der Matrix sind die Impedanzen Z Kernimpedanzen, d.h. bestimmte, für das System charakteristische Impedanzausdrücke. Die Kernimpedanzen stimmen bei den einfachsten akustischen Elementen mit den Widerständen nach den Gln. (1) und (2) überein. Darüber hinaus drückt die Matrix aber auch die Übertragungseigenschaften aus. Mit einfachen Rechenregeln beherrscht man die Zusammenschaltung mehrerer Elemente, die Eigenfrequenzen usw.

Das elektrische Analogieschaltbild folgt unmittelbar aus der Matrix. Sollen akustische Impedanzen durch elektrische Widerstandselemente dargestellt werden, so zeichnet der Elektrotechniker auf grund der Matrix folgende Schaltungen an:



Abb. 2. (a) Hohlraum, (b) Rohr.

Sollen akustische Impedanzen durch elektrische Leitwertselemente dargestellt werden, so sind die Analogieschaltbilder die dualen:



Abb. 3. (a) Hohlraum, (b) Rohr.

Die Analogieschaltbilder zeigen, daß es sich auch bei den einfachsten akustischen Elementen um Vierpole handelt, und zwar um sogenannte entartete Vierpole. Für analoge Zweipole nach den Gleichungen (1) und (2) bedarf es ergänzender Schaltungsvorschriften.

SECTION IV

AUDIOMÉTRIE ET APPAREILS DE PROTHÈSE

REVUE GÉNÉRALE

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La Haute Autorité du Congrès de Delft nous a dévolu la passionnante, mais délicate mission d'étudier les questions relatives à l'audiométrie et aux appareils de prothèse: c'est-à-dire en bref, de traiter de ce qui concerne l'un des dons les plus précieux que nous ait faits la création: je veux dire "l'ouïe". C'est un problème passionnant, infiniment intéressant, car l'expérience quotidienne nous démontre la grande misère des malentendants. Rappellerais-je simplement, comme nous le faisions lors du récent Cours International d'Audiologie Clinique, l'opinion d'un de nos vieux humanistes français, MONTAIGNE, que je m'excuse de reprendre dans cet exposé, mais elle apporte une note subjective, sensible et humaine, dans un domaine où la passion scientifique peut risquer de nous entraîner vers des recherches trop abstraites et nous faire oublier le but final de notre mission qui est de mieux faire connaître les lois auxquelles obéit le mécanisme de nos sensations, tout en contribuant à soulager efficacement ceux qui souffrent dans leur deuxième sens. Voici ce qu'on lit, en effet, dans l'Art de conférer (Essais, Livre III, Chap. 8): "Le plus fructueux et naturel exercice de notre esprit est à mon gré la conférence (conversation). J'en trouve l'usage plus doux que d'aucune autre action de notre vie et c'est la raison pourquoi, si j'étais à cette heure forcé de choisir, je consentirais plutôt, ce crois-je, de perdre la vue que l'ouïe ou le parler."

Mais, à peine nous sommes-nous penchés sur ces problèmes de l'audition et de la correction auditive, que nous nous apercevons de leur complexité, de leur difficulté. L'une et l'autre suffisent à expliquer la lenteur avec laquelle ont progressé les recherches qui leur ont été consacrées. Il faut, d'ailleurs, reconnaître que si, suivant un principe fameux de la philosophie des sciences, il n'y a de science que de mesurable, nous ne nous intéressons vraiment à une recherche, à une science, que lorsque nous voyons transparaître à travers l'aridité des théories et des équations qui les définissent, l'intérêt intellectuel d'une satisfaction de l'esprit, certes, mais aussi l'espoir humain, plus humble mais parfois plus générateur d'efforts, d'apporter quelques soulagements à notre condition humaine.

Or, l'audiométrie et la prothèse auditive ont, n'est-il pas vrai, des objets, des ambitions qui répondent à ce double appel de notre esprit et que je résume ainsi:

— mieux connaître l'audition, donc la mesurer avec plus de finesse, plus de rigueur aussi et à cette fin, préciser la technique audiométrique;

— ayant, grâce à cette technique, constaté, délimité les insuffisances de l'audition, y remédier dans la mesure du possible: c'est l'objet de la prothèse auditive.

Indiquons les grandes lignes des chemins parcourus par savants et techniciens, physiciens et médecins, dans cette double direction, avec la double ambition, la double préoccupation de la science pure (les vérités générales et la connaissance en soi) et de la technique appliquée aux cas particuliers, c'est-à-dire à la vie.

Voulez-vous qu'avant d'apprendre de nos collègues les révélations de leurs derniers progrès scientifiques, jetant un regard vers le passé, nous esquis-

sions un rapide historique des "premiers faits" audiologiques et de leur évolution. Regard très bref si nous ne voulons pas nous attarder puisque, à lui seul, le Cours d'Audiologie Clinique professé à Paris du 11 au 17 février 1952 n'a pas réuni moins de 23 conférenciers dont les travaux forment un volume de 600 pages. Sa rapidité même nous permettra, peut-être, de mieux mettre en lumière les remarquables progrès que nous devons à une floraison de travaux internationaux, dont la possibilité et le développement ont un point de départ commun à travers les progrès de l'électroacoustique et de la téléphonie, celui de la découverte des phénomènes électroniques et de la lampe à trois électrodes par Lee de Forest.

Historique de l'audiométrie

Si l'audiométrie a une très vieille histoire, axée sur le phénomène même de l'audition, elle le doit à l'observation des faits spécifiques qui signalaient la mauvaise compréhension des durs d'oreille. La difficulté de comprendre la voix parlée, l'impossibilité d'entendre les sons faibles (celui du diapason dont les vibrations s'éteignent lentement après percussion initiale, tic-tac de la montre que l'on éloigne de l'oreille), ont conduit les otologistes à caractériser le degré d'aptitude auditive et la déficience auditive par une appréciation mi-qualitative, mi-quantitative: qualité de la voix perçue ou entendue, voix chuchotée, voix murmurée, voix mi-forte, voix très forte, temps d'audition d'un diapason par conduction aérienne à une distance donnée de l'oreille ou par conduction osseuse, distance minima de perception du tic-tac de la montre.

Si je crois devoir rappeler ces procédés qui paraissent désuets et dignes des premiers âges, c'est que des études récentes ont souligné à "l'évidence" qu'ils présentaient un réel intérêt quantitatif. Tel était bien, en effet, la pensée de notre ami le Docteur Raoul Caussé lorsque nous fûmes conduits, il y a quelques années, à préciser les possibilités audiométriques du test à la montre. Quant aux essais d'intelligibilité dont les téléphonistes ont montré, il y a longtemps, l'importance dans l'évaluation de la qualité de la transmission, de la conversation téléphonique, ils ont trouvé, ainsi que l'ont remarqué de nombreux auteurs, un champ d'application extrêmement étendu dans l'évaluation de la surdité et de la correction auditive. Ainsi, dans cette science, comme dans bien d'autres, des idées élémentaires primitivement à peine ébauchées ont révélé leur fécondité quand elles ont trouvé, pour les supporter et les mettre en valeur, les fondements ou les moyens scientifiques qui leur faisaient défaut.

Grâce à Harvey Fletcher, l'étude de toutes ces questions, de leurs diverses modalités et de leurs possibilités techniques, reçut une impulsion essentielle qui les recréa, en quelque sorte, et qui leur donna un développement scientifique qui n'a cessé de progresser jusqu'à nos jours. L'audiométrie est, en effet, scientifiquement née avec les recherches de Wegel, de Lane et surtout de l'auteur de "Speech and Hearing". J'ai cité ces quelques noms que tous connaissent d'une façon presque classique et qui ont réellement marqué la science de l'ouïe; cependant, je préfère que nous nous en tenions maintenant aux faits plus qu'aux noms, car dans la floraison des travaux qui ont jalonné la progression de l'audiologie, il y eut tellement d'apports originaux, illustres ou plus obscurs, de constructions nouvelles, que la crainte de laisser de plus dignes dans l'ombre nous conseille de nous attacher avant tout aux réalités qui ont marqué les stades successifs des progrès de l'audiométrie jusqu'à l'état où nous la voyons aujourd'hui, avec une perspective toute grande ouverte sur les révélations de l'avenir.

Avant la guerre, peu de règles avaient pu être élaborées. Notons, cependant, une tentative américaine et un examen comparatif inquiétant des audiomètres communément utilisés, examen montrant leur extrême individualisme et la diversité fantaisiste des audiogrammes qu'ils permettaient de tracer; de même, en 1938, un projet français où seuls étaient précisées les caractéristiques électriques des audiomètres, de manière à garantir la construction et la qualité matérielles; mais, faute d'oreille et de mastoïde artificielles et de chambre muette, faute enfin de zéro absolu, aucune construction définitive ne put être édifiée. Du moins, ces premières tentatives permirent-elles de poser correctement le problème, de voir avec précision leurs difficultés, de les classer en quelque sorte et d'orienter vers leurs solutions méthodiques.

Ceci nous conduit au problème majeur de l'audiométrie, celui du zéro de référence, c'est-à-dire du seuil normal aérotympanique de l'audition, et ici je laisse volontairement de côté le problème beaucoup plus complexe encore du seuil normal par voie osseuse.

La définition correcte du seuil aérotympanique suppose, en effet, une convention: car, en définitive, qu'est-ce qu'un seuil d'audition? Subjectivement, nous n'avons aucune peine à le définir par une sensation liminaire et, physiquement, par une intensité ou une pression acoustique, donc, par des éléments mesurables. Or, dès que ce problème fut posé, il apparut d'une extrême difficulté à résoudre: qu'on opérât avec un écouteur téléphonique placé contre le pavillon de l'oreille ou dans le conduit auditif, qu'on exécutât les mesures avec un haut-parleur, il fallait de toutes façons, une convention. Des travaux comme ceux de Waetzmann et Keibs, de Sivian

et WHITE en 1933, de MONTGOMERY en 1932, de BEASLEY en 1936, de DADSON et KING et de WHEELER et DICKSON en 1952, ont montré des écarts considérables entre les deux conceptions. Ainsi, voyait-on s'esquisser deux tendances, deux techniques, deux écoles, deux définitions du zéro: celle des seuils avec écouteur, celle des seuils avec haut-parleur.

La première technique, relative au seuil avec écouteur, a conduit les expérimentateurs à examiner en détail le fonctionnement des écouteurs et leurs conditions de rayonnement et à délimiter avec précision le problème de la définition du seuil. De la sorte, on fut incité à étudier l'impédance acoustique de l'oreille naturelle, à définir et à construire des oreilles artificielles mécaniquement et physiquement spécifiées, qui soient représentatives d'une oreille moyenne typique.

Rappelons ici que divers modèles d'oreilles artificielles ont été réalisées dès 1932 en Amérique (INGLIS, GRAY, JENKINS), puis dans la suite en Angleterre (WEST), en Italie (BARDUCCI), en Suisse (WEBER) et en Allemagne; en France également des études de même nature ont conduit à des créations du même genre (au Centre National d'Études des Télécommunications). Des comparaisons entre les divers modèles ont fait apparaître une certaine "homogénéité" dans les résultats des essais relatifs à des écouteurs dont la réponse avait été relevée avec diverses oreilles, ce qui ouvre des perspectives très favorables sur la définition des seuils. Cette normalisation sera nécessaire et permettra, sans doute, de ramener au même niveau les relevés systématiques de seuils déterminés dans divers pays sur quelques centaines, quelques milliers ou quelques centaines de milliers de sujets, tant en Amérique qu'en Angleterre, qu'en Italie, en Suisse, et en France.

En ce qui concerne les essais avec haut-parleur qui imposent l'expérimentation en salle muette, ils ont eu de nombreux protagonistes et il n'est que de citer ici les publications de SIVIAN et WHITE et de FLETCHER et MUNSON pour illustrer leur rare qualité. Sans prononcer d'autres noms, rappelons que le grand problème est celui que pose l'effet de diffraction de la tête, la nécessité de préciser la forme des ondes et l'orientation de la tête par rapport à la direction de la propagation, ainsi que l'influence du conduit auditif. Enfin, dans le cas de l'écoute en onde libre, apparaissent tous les phénomènes inhérents au mécanisme binauriculaire (augmentation de la sensibilité auditive, sensibilisation temporaire, nécessité de masquer une des oreilles pendant que l'autre est examinée, etc.).

On sait que des études comparatives ont été faites entre les valeurs des seuils rapportés au tympan ou avec écouteur et les valeurs relevées en champ libre (WIENER, ROSS, MUNSON entre autres), et que des écarts de l'ordre d'une dizaine de décibels ont été constatés entre les unes et les autres.

On se souvient aussi que, compte-tenu des effets de diffraction ou d'impédance acoustique, il subsiste encore des différences entre les seuils déterminés par ces divers genres de mesures, divergences qui, en 1953, semblent à notre connaissance encore inexpliquées.

Ceci laisse entrevoir l'intéressante perspective de recherches fructueuses et indique que tout n'a pas été dit en matière de seuil d'audition et que la voie reste largement ouverte aux prospections scientifiques. Retenons simplement, si vous le voulez bien, les questions qui préoccupent tous les techniciens et qui tracent la route des futurs travaux:

— écoute en champ libre: spécification des salles d'écoute, des assourdisseurs, effet de l'orientation de la tête, etc.;

— examen avec écouteur: spécification d'une oreille artificielle ou du moins de ces caractères physiques généraux;

— dans les deux cas d'écoute, application des notions nouvelles concernant le recrutement, les seuils différentiels, la constante de temps de l'oreille, la fatigue auditive;

— définition d'une mastoïde artificielle pour l'examen des vibrateurs (v. BÉKÉSY, HAWLEY, GREIBACH, CARLISLE . . .).

Plus généralement, définition de normes universelles permettant aux otologistes de caractériser de la même manière un déficit auditif, de parler un langage uniforme, de se comprendre et de s'entendre entre eux, ce qui semble le minimum que l'on puisse requérir de ceux qui ont pour mission de faire entendre les autres.

Appareils de prothèse

Le deuxième chapitre, complément, conséquence et conclusion du premier, est celui de la prothèse auditive. C'est en définitif celui qui intéresse le plus le patient auquel il apporte quelqu'apaisement. La technique des appareils, qui a beaucoup évolué sur le plan pratique a, jusqu'ici, consisté en une amplification indistincte et uniforme des sons et tout spécialement des sons vocaux.

Il est évident que leur technique a suivi l'évolution générale de l'électronique, alors qu'ils étaient demeurés longtemps au stade des antiques porte-voix et des cornets acoustiques, dont le rôle était de conduire à l'oreille les sons de la parole sans leur faire perdre de leur intensité très considérable au voisinage des lèvres: quelques 100 phones. Ne parlons ici que pour mémoire, des premiers appareils amplificateurs qui étaient basés sur le pouvoir amplificateur du microphone à charbon et dont la qualité, souvent mauvaise et irrégulière, a découragé nombre

de malades. La création des tubes électroniques et des appareils de haute qualité: microphone piézoélectrique, récepteur magnétique, a dans ce domaine, comme dans beaucoup d'autres, amené une "révolution créatrice". Elle a été complétée par l'apparition des vibrateurs de LIEBER et de leurs dérivés qui ont permis de corriger des surdités qui n'étaient pas justifiables des anciennes méthodes d'amplification.

La révolution dans l'appareillage a été accélérée lors de la récente guerre par l'apparition des tubes miniatures, des piles de longue conservation et de petites dimensions. Enfin, l'application générale des méthodes de l'électronique a permis, par la mesure des seuils d'audition, des pouvoirs amplificateurs et par le tracé des courbes de réponse (utilisant encore ici les oreilles artificielles) de définir les propriétés des appareils, de les chiffrer, d'en évaluer le bénéfice et d'orienter les progrès futurs.

Se sont ainsi posés successivement: le problème de l'amplification uniforme de haute qualité, tant en conduction aérotympanique qu'en conduction osseuse, puis celui de l'amplification sélective, enfin celui de la régulation automatique du volume, si précieuse dans de nombreuses affections, mais tout spécialement dans celles qui se compliquent du phénomène de recrutement. De toute façon, l'appareillage, évidemment très recherché des sourds, est maintenant fourni par les services de la Sécurité Sociale, mais en contrepartie, un contrôle a dû être imposé.

Ce contrôle a porté sur la qualité, la stabilité, la définition des courbes de réponse qui, à l'image des courbes de réponse des appareils téléphoniques, ont pu être relevées au moyen d'oreilles artificielles, de bouches artificielles et de mastoïdes artificielles. Citons également les études relatives à l'effet de baffle humain (HANSON, NICHOLS . . .).

Ces appareils ont permis, d'une part, de normaliser, au moins provisoirement, certaines courbes de réponse jugées jusqu'ici les mieux adaptées à la correction auditive dans la majorité des cas. Telle est la solution adoptée dans divers pays, notamment en France. Mais ils ont ouvert les perspectives nouvelles en permettant, par les méthodes d'essais subjectives d'intelligibilité mises au point par divers auteurs (FOURNIER, FALCONNET etc.) d'établir une corrélation entre la forme d'une correction et sa qualité. Ainsi s'est trouvé posé le problème de la correction sélective, problème difficile, controversé, mais essentiel (DAVIS, STEVENS, . . .).

Ici, Messieurs, je m'arrête sans tirer de conclusion précise, car la conclusion, ce sont vos travaux, vos débats qui la susciteront, qui l'éclaireront, qui l'imposeront. En ce qui concerne le problème de l'appareillage, de grands progrès ont été faits qui autorisent les plus grands espoirs. Si les méthodes de contrôle avec voix, bouches, oreilles artificielles ont pris un caractère universellement scientifique, en revanche un accord international reste encore à obtenir sur divers points et notamment sur les suivants:

1) Courbes de réponse optimum de l'appareil de prothèse auditive;

2) Leur adaptation aux divers types de surdité;

3) L'oreille et la mastoïde artificielles à utiliser pour l'étude des appareils;

4) Problème de la régulation automatique du volume.

C'est-à-dire, au fond, création d'une norme générale de caractère international, norme aux spécifications harmonieusement diversifiées tout en restant dans le cadre d'une unité fondamentale.

J'ajouterai, Messieurs, union de vos efforts, de vos études théoriques et pratiques, physiques ou subjectives, comparaison des résultats dont il se dégagera, j'espère un jour, des solutions concrètes, indiscutables qui s'imposeront à l'esprit des techniciens. Ce jour-là, nous aurons utilement travaillé. Permettez-moi, Messieurs, d'espérer que cette réunion de Delft aura de tels effets pour le plus grand bien de nos frères.

A DISCUSSION OF SOME PROBLEMS INVOLVED IN DERIVING OBJECTIVE PERFORMANCE. CRITERIA FOR A WEARABLE AID FROM CLINICAL MEASUREMENTS WITH LABORATORY APPARATUS

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Summary. In investigating the effects of such parameters as frequency-response characteristics on the performance of hearing aids, recourse is often necessary to experiments under laboratory conditions. In translating the results of such experiments into a performance specification for a wearable aid, it is necessary to make allowances for differences in the relationships between the subjective and objective efficacies of the transducers used, and for the special conditions of use of the hearing aid. The microphone of the aid is normally worn on the body of the user, possibly under one or more layers of clothing, and is exposed to a partially diffuse sound field. The paper discusses some of these effects, and shows how the appropriate correcting factors may be determined.

Sommaire. Dans une investigation des influences de paramètres comme les courbes de fréquence pour les appareils de prothèse, on doit souvent recourir à des expériences de laboratoire. Pour obtenir une spécification pour un appareil de prothèse portatif il est nécessaire d'avoir égard aux efficacités subjectives et objectives des transducteurs employés et aux conditions spéciales de son emploi. Le microphone de l'appareil est porté sur le corps, peut être sous un ou deux vêtements de l'usager et est exposé à un champ sonore partiellement diffus. On discute certaines de ces influences et on montre comment les corrections peuvent être déterminées.

Zusammenfassung. Zur Beurteilung der Wirkungsweise von Hörapparaten werden oft Versuche unter Laboratoriumsbedingungen herangezogen. Um hieraus Vorschriften für die Leistung tragbarer Apparate zu gewinnen, muß man die Unterschiede zwischen der subjektiven und objektiven Wirksamkeit des Gerätes und die besonderen Benutzungsbedingungen berücksichtigen. Einige Auswirkungen — z.B. wenn das Mikrophon unter Kleidungsstücken getragen wird — werden besprochen und es wird gezeigt, wie entsprechende Korrekturfaktoren bestimmt werden können.

1. Introduction

This paper is concerned with the gain and output power of hearing aids, as functions of frequency. I do not propose to re-open the controversial question of optimum characteristics, but to discuss a problem which confronts all who tackle it in a scientific way — the problem of translating clinical measurements under laboratory conditions into an objective performance specification for a wearable aid.

2. Subjective equivalence

Let us assume that measurements, for example of intelligibility, have been made with a "master hearing aid", consisting of a laboratory microphone used in an acoustically treated room, amplifiers, equalisers, etc., and a high quality receiver. It is immaterial for the present whether the measurements are of one individual or of a representative group, or what criterion is used in judging results, but we may assume that certain settings of this equipment have been found to give a desired performance, which is to be reproduced in a wearable hearing aid. The preferred settings of the master aid may be described objectively, for example by the gain function:

$$A(f) = 20 \log (P_1/P_0) \text{ dB} \qquad (1)$$

where P_1 is the sound pressure developed, in a specified artificial ear, by the master aid receiver when its microphone is exposed to a plane progressive sound field of unobstructed sound pressure P_0.

It will be assumed that the subjective effect of the master aid at this setting will be reproduced in a wearable aid if, from a given sound source, the latter is arranged to develop at each frequency the same loudness as the master aid would have developed. The relationship between objective and subjective effects depends on the types of transducer employed and the way in which they are used, so that it is not sufficient merely to duplicate the A function. For example, a given sound pressure may be set up in the ear canal by a free sound field or by various types of closely coupled receivers, but it does not follow that the resulting loudness sensations will be equal. The differences are particularly large with insert type receivers. It is necessary also to allow for the fact that a wearable aid microphone is commonly not used in the same way as the master aid microphone, i.e. suspended in a free sound field with no other obstruction near. The aid is worn on the body, possibly under one or more layers of clothing, and is exposed to a partially diffuse sound field.

3. Receiver corrections

The objective sensitivity of the master aid receiver may be defined as:

$$S_o(f) = 20 \log (P_1/E_1) \qquad (2)$$

where E_1 is the electrical input required to develop the sound pressure P_1 in the specified artificial ear. If the receiver of the wearable aid requires an input E_2 to develop P_1 in an appropriate artificial ear, not necessarily the same, its sensitivity is:

$$T_o(f) = 20 \log (P_1/E_2). \quad (3)$$

Subjective calibrations may now be made to find the free-field sound pressures P_2, P_3 required to give the same loudness sensations as produced by the receivers when excited by E_1, E_2:

$$S_s(f) = 20 \log (P_2/E_1), \quad (4)$$

$$T_s(f) = 20 \log (P_3/E_2). \quad (5)$$

The target function A may now be expressed as a ratio of free-field sound pressures only. P_1 in (1) may be replaced by the equivalent free-field sound pressure P_2, using (2) and (4), giving

$$\begin{aligned} G(f) &= 20 \log (P_2/P_0) \\ &= 20 \log (P_1/P_0) + 20 \log (P_2/E_1) - 20 \log (P_1/E_1) \\ &= A + S_s - S_o \text{ dB}. \end{aligned} \quad (6)$$

This may be described as the target monaural free-field insertion gain of the master aid. Under the assumptions made in the previous section, subjectively equivalent systems must produce the same loudness from a given sound source, so G is also the target for the wearable aid *if used with a free field input*. By reversing the process leading to (6), an objective gain function for the wearable aid, analogous to A, may be deduced, giving the pressure P_4 which should be developed by the wearable aid receiver, in the artificial ear specified for its calibration, when its microphone is exposed to a free field of unobstructed sound pressure P_0.

$$\begin{aligned} B(f) &= G - (T_s - T_o) \\ &= A + (S_s - S_o) - (T_s - T_o) \\ &= 20 \log (P_4/P_0) \text{ dB}. \end{aligned} \quad (7)$$

This quantity B, though convenient for objective measurement, fails to allow for the proximity of the body, or for use of the aid in reverberant surrounings.

4. Body baffle effect

Attemps have been made to devise artificial bodies, to which an aid may be attached when immersed in a free sound field for test, and which simulate the absorbing and diffracting effects of an average wearer. Unfortunately neither artificial nor real human bodies are satisfactory for this purpose under the laboratory conditions of a plane progressive incident sound field. Fig. 1 illustrates the rapid variation with frequency and subject of the pressure at the middle of the chest. Fig. 2 taken from data published by Harvard University [1] shows the effect of the body on the output of a number of aids. The various curves in this figure show apparently unrelated fluctuation, and the authors quote large changes as a result of spacing the aid only a centimetre or so from the surface of the body. The problem of producing a representative artificial body appears to be a formidable one, but I will return to this after considering the question of reverberant surroundings.

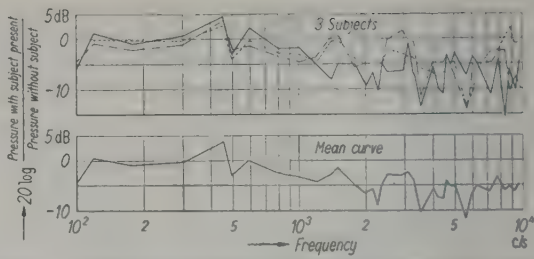


Fig. 1. Change in pressure at a point in a free sound field due to introducing thereat the chest of a human subject.

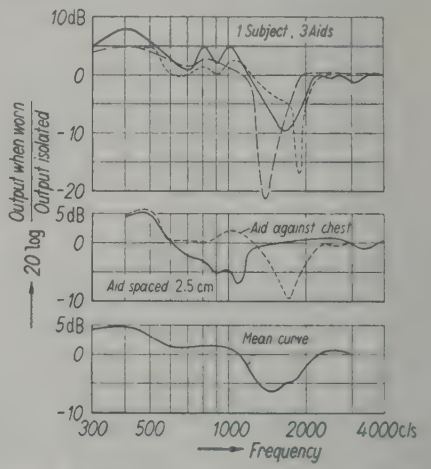


Fig. 2. Body-baffle effect in a free sound field.

5. Diffuse fields

It may perhaps be assumed that the progressive sound field used in calibrating the master aid is a sufficiently close approximation to its normal conditions of use. If so, the quantity G of (6) adequately describes its performance. To allow for the different conditions of use of the wearable aid, the definition of subjective equivalence, equal loudness for a given input P_0, must now be broadened to equal loudness for an equivalent input, i.e. a sound field of *random* incidence with an unobstructed sound pressure numerically equal to P_0. With the understanding that P_0 refers to such a random field, the quantity B now represents the target gain function for the wearable aid, *measured in a diffuse sound field and attached to an artificial body*.

Returning now to the question of the artificial body, Fig. 3, from the same source as Fig. 2, shows

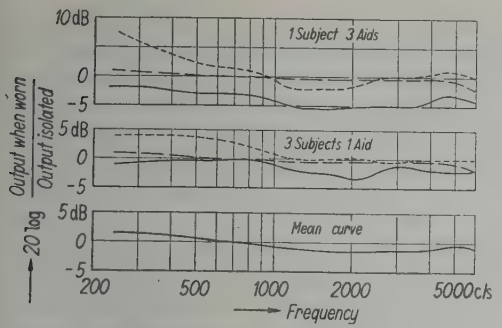


Fig. 3. Body-baffle effect in a random sound field.

the effect of the body on the output of an aid in a diffuse field. The behaviour is much more consistent than in progressive sound fields, and a stable mean curve can be drawn. This may either be used to specify an artificial body, or applied as a correction factor to measurements made with an isolated aid. If

$$a(f) = 20 \log \frac{\text{mean output when worn}}{\text{output when isolated}}, \quad (8)$$

a new target function may be defined for the measured gain of the *isolated* aid in a diffuse field:

$$B'(f) = A + (S_s - T_s) - (S_o - T_o) - a \text{ dB}. \quad (9)$$

Finally, since it is not always convenient to set up truly random testing fields, the performance of an isolated aid in such a field may be estimated by calculating the ratio of its mean spherical to normal sensitivity from measurements with progressive waves incident at a number of angles. If

$$b(f) = 20 \log \frac{\text{mean spherical sensitivity}}{\text{normal sensitivity}} \text{ dB} \quad (10)$$

and the isolated aid is now tested with a plane progressive sound wave, P_0, at normal incidence, the sound pressure produced by its receiver in an artificial ear, to meet the target conditions, should be P_5 and its gain should be B'', given by:

$$B''(f) = 20 \log (P_5/P_0)$$
$$= A + (S_s - T_s) - (S_o - T_o) - a - b \text{ dB}. \quad (11)$$

Reference

[1] Nichols, R. H. et al., J. acoust. Soc. Amer. 19 [1947], 943.

METHODS OF MEASUREMENT FOR HEARING AIDS

by A. G. Th. Becking

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Summary. The different ways of defining the input sound pressure on a hearing aid are discussed. In addition, a comparison is given of the specifications for acceptable hearing aids in some European countries.

Sommaire. Étude des méthodes définissant la pression acoustique d'entrée pour les appareils de prothèse auditive. En outre, est présentée une comparaison des spécifications valables dans quelques pays Européens.

Zusammenfassung. Die verschiedenen Methoden für die Festlegung des Eingangsschalldrucks bei der Messung von Schwerhörigengeräten werden diskutiert. Außerdem werden die in verschiedenen europäischen Ländern gültigen Normen für Schwerhörigengeräte miteinander verglichen.

1. Introduction

A great deal has already been written and said about methods of measurement for hearing aids. But the subject is still a source of many misunderstandings and it seems that no two persons measure the hearing aids in exactly the same way. Perhaps this can serve as an excuse for one more paper, even if some of the ideas it contains have already been put forward [1]. We will restrict ourselves to one point in the procedure; viz. the way in which we define and measure the input sound pressure which acts on the microphone of the hearing aid.

2. Input sound pressure definitions

The method most quoted is the free-field method (e.g. [2]). A progressive wave is produced in an anechoic room and the sound pressure in it is measured. After this the hearing aid is inserted at the measuring spot. This corresponds to the usual definition of the field-calibration of standard microphones.

Another method is more like the pressure-calibration definition. Here the sound pressure, as actually present at the microphone openings of the aid, is regarded as the input sound pressure. To measure it, a standard microphone is brought as close as possible to these microphone openings in the aid. It is often convenient to use this standard microphone as a regulating microphone, thus keeping the input sound pressure constant. This measurement can be done in an anechoic room but it is also possible to do it in a small box.

These two definitions — "field" and "pressure" — can give quite different results. This can be shown by the following experiment (Fig. 1). A condenser microphone is inserted into a progressive wave in

an anechoic room. This microphone is of such small dimensions that it does not disturb the sound field at the frequencies that interest us.

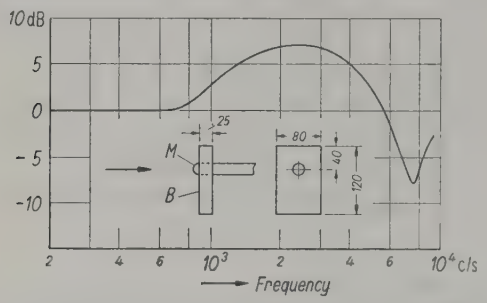


Fig. 1. Change in response of the microphone M when the block B is put on it. The arrow gives the direction of the incident sound; the dimensions of the block are given in mm.

A piece of wood in the form of a hearing aid is then slipped over the microphone. The change in response of the microphone (Fig. 1) is then just the difference between field and pressure calibration of a hearing aid of the same form as the block. This block is somewhat larger than present-day aids, but this can be accounted for by multiplying the frequency scale by a factor of 1.5 or 2. Even then there remain appreciable differences in the speech range up to 4000 c/s.

Fig. 2 shows that when we keep the sound pressure near the "aid" constant by means of a separate regulating microphone, we return to the pressure calibration.

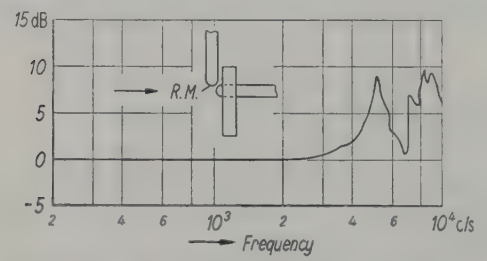


Fig. 2. The same block is used as in Fig. 1, but now the sound pressure as measured by the regulating microphone R.M. is kept constant.

An argument in favour of the pressure method is its smaller cost, as no anechoic room is needed. This is important, as it is useless to adopt a very fine experimental method when its cost will limit its use to only a few laboratories.

3. The body-baffle effect

When the hearing aid is in actual use, the incoming sound waves will be diffracted by the body of the patient. The question is now to decide which of the two, namely the field or the pressure characteristic, will enable us to predict the behaviour on the patient most accurately. To answer this question, the experiment of Fig. 1 was repeated with a flat baffle (imitating the body of the patient) (Fig. 3), and with the block on the baffle (Fig. 4, curve drawn in full). Only in the latter case does the familiar body-baffle dip appear, and we see that a new diffraction curve results, which is totally different from both the curve for the aid alone and the curve for the baffle alone. Apparently neither field nor pressure calibration can predict for us the true characteristic when the aid is worn on the body.

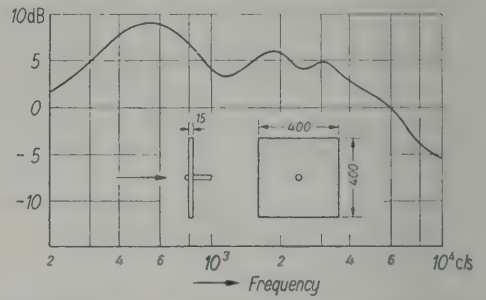


Fig. 3. The experiment of Fig. 1 repeated with a baffle.

There is a great need for international standardisation of methods of measurement for hearing aids. In the writer's opinion it would be useful to normalize both the field and the pressure method: the first because it is already widely accepted, the second because of its simplicity.

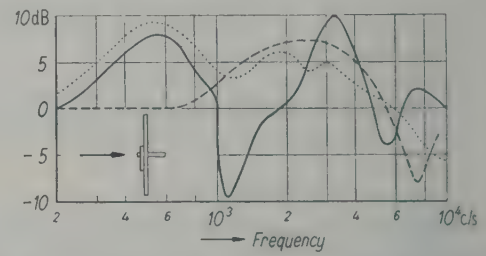


Fig. 4. The block of Fig. 1 is placed before the baffle of Fig. 3, simulating a hearing aid worn on the body of the user (curve drawn in full).
For comparison are added:
— — — — obstacle effect of hearing aid alone, as in Fig. 1.
. . . . . obstacle effect of baffle alone, as in Fig. 3.

4. Specifications on overall curves

In some European countries there are organisations that have drawn up specifications for hearing aids. These specifications are as different as the experimental methods are. As an example we shall compare the overall acoustic curves required by two organisations in two different countries. They will be referred to as A and B.

In case B, the ASA type 2 (2 cm³) coupler is used; in case A, a different artificial ear. When a correction (6 dB/octave up to 2000 c/s) is applied for this, the resulting tolerance limits are as shown in Fig. 5. The language difference between A and B can hardly justify this divergence.

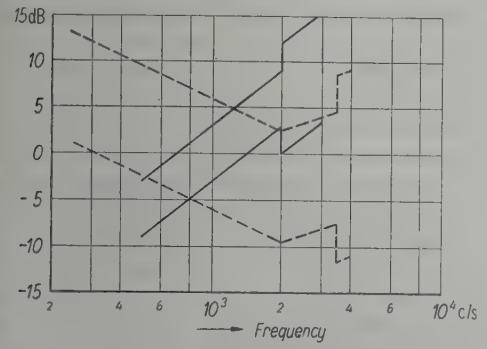


Fig. 5. — — — — Tolerance limits in case A.
———— Tolerance limits in case B.

Both specifications can yield a hearing aid that is satisfactory for a certain category of patients. The solution of the standardisation problem should be sought in the specification of *a number* of curves, each for a type of deafness, as was put forward by CHAVASSE and LEHMANN [3].

References

[1] CORLISS, E. L. R. and COOK, G. S., J. acoust. Soc. Amer. 20 [1948], 131.
[2] American Hearing Aid Association, J. acoust. Soc. Amer. 17 [1945], 144.
[3] CHAVASSE, P. et LEHMANN, R., Acustica 2 [1952], 119.

Discussion

AYERS, E. W.: Pressure calibration of aids fails to show effects of sound reaching the microphone through the back of the case, e.g. The response may appear good with pressure calibration, but will be irregular with field calibration.

LECTURER: When the aid is worn, the back openings are covered by clothing. Even if these openings would then still transmit sound, conditions will be different also from those during free field calibration.

MRASS, H.: Wird das Regelmikrophon 2 bis 5 cm seitlich von der Hörhilfe angebracht, so ergeben sich gegenüber einer Freifeldmessung keine größeren Abweichungen als ± 3 dB. Dies ist von geringer Bedeutung.

LECTURER: Diese Meßmethode ist sicherlich möglich und gibt eine Art Mittelwert zwischen Feld- und Druckeichung. Wenn man aber von der Feldmethode abgeht, ist es wohl prinzipiell besser, dann auch gleich zur reinen Druckmethode überzugehen.

VALEURS COMPARATIVES DES DÉFICITS AUDITIFS MESURÉS AVEC UN BRUIT COMPLEXE ET AVEC DES SONS PURS

par P. CHAVASSE et P. FALCONNET

Centre National d'Études des Télécommunications, Paris. Centre d'Examen Médical du Personnel Navigant de l'Aviation, Paris

Sommaire. Le problème considéré par les auteurs est celui de la substitution, en audiométrie, d'un test unique avec bruit complexe, aux tests multiples habituels avec fréquences pures, dans une bande de fréquences déterminée, et de rechercher s'il existe une corrélation entre les résultats donnés par les deux méthodes d'investigation. En l'absence de définition valable de l'intensité du bruit complexe à spectre continu, les mesures subjectives effectuées ont toutes été relatives: elles sont rapportées aux seuils de sujets normaux.

Zusammenfassung. Die Autoren behandeln das Problem, die in der Audiometrie üblichen Tests mit vielen reinen Tönen durch einen einzigen Test mit einem komplexen Geräusch in einem bestimmten Frequenzband zu ersetzen und versuchen, eine Beziehung zwischen den mit beiden Methoden gewonnenen Resultaten zu finden. Ohne eine genaue Definition für die Energie des komplexen Geräusches mit kontinuierlichem Spektrum war man bei subjektiven Tests auf Relativmessungen in Bezug auf die Hörschwellen normaler Versuchspersonen angewiesen.

Summary. The problem considered is that of the substitution in audiometry of a single test with a complex noise for the multiple tests commonly made with pure tones in a given frequency band, and of finding whether a correlation exists between results obtained with the two methods. In the absence of a valid definition of the intensity of the white noise, the subjective measurements made were all relative. These related to the thresholds for normal subjects.

1. Introduction

Sur le plan physio-pathologique, les bruits complexes ont été surtout envisagés jusqu'à ce jour en fonction de leur effet de masque. Mais il nous est apparu depuis longtemps qu'une application intéressante de bruit complexe à spectre continu ou plutôt, des bruits complexes obtenus par introduction de filtres de bandes, pouvait se trouver dans la recherche des seuils de l'audition à partir de tels bruits en remplacement des sons purs. Une telle méthode aurait pour intérêt d'explorer l'oreille, en même

temps, sur toute une gamme de fréquences et présenterait de multiples avantages.

L'hypothèse du présent travail résidait dans cette déduction, plausible, que le seuil de l'audition d'un bruit complexe, dans une bande de fréquences déterminée, correspond au seuil de la fréquence pure la mieux perçue dans cette bande. En d'autres termes, l'oreille, interrogée simultanément sur un certain nombre de fréquences, répondrait dès qu'elle entend l'une de ces fréquences, soit la fréquence dont le seuil est le plus bas, c'est à dire le meilleur.

2. Description de l'appareillage

Les essais que nous rapportons ici ont été effectués à l'aide d'une source de bruit complexe produit par agitation électronique dans un gaz ionisé. Les fluctuations de la tension prélevée aux bornes du tube à gaz sont appliquées à l'entrée d'un amplificateur dont la sortie est connectée à l'entrée des filtres.

Le montage de l'appareillage, tout à fait classique, comprend, entre le générateur de bruit et la ligne d'affaiblissement qui précède l'écouteur, deux filtres de bandes, le premier laissant passer les fréquences supérieures à 400 Hz et le second celles inférieures à 1900 Hz. Avec un analyseur du type spectromètre de bandes, tel que celui de Siemens, spectromètre à $\triangle F/F = C^{te}$, (c'est-à-dire, un appareil dont la sélectivité absolue $\triangle F$ est proportionnelle à la fréquence, la selectivité relative $\triangle F/F$ étant la même dans toute l'etendue du spectre) le bruit complexe 400 ··· 1900 Hz, issu du montage, présente, à l'analyse spectrale, une distribution linéaire, soit une intensité relative à peu près égale dans chacun des intervalles élémentaires de fréquences de la bande considérée.

On utilisait, en outre, un générateur de sons purs relié à un écouteur de haute qualité par l'intermédiaire d'une ligne d'affaiblissement permettant les mesures audiométriques habituelles.

3. Mesures effectuées

Avec chacun des deux appareillages, à bruit complexe ou à fréquences pures, la tension de mesure, repérée au moyen d'un voltmètre à oxyde de cuivre, était maintenue constante à l'entrée de la ligne d'affaiblissement. La détermination du seuil normal de l'audition, à l'aide de l'appareillage à fréquences pures avait été préalablement effectuée, pour des essais d'une autre nature, sur 56 oreilles saines. On avait ainsi mesuré la valeur moyenne de l'affaiblissement qu'il convenait de donner à la tension de sortie du générateur pour obtenir les seuils des fréquences 500, 800, 1000 et 2000 Hz. L'appareillage à bruit complexe a été pareillement étalonné, mais en même temps que les essais ici rapportés, sur 32 oreilles normales ayant servi à une partie des mesures effectuées dans la suite de cette communication, et l'on avait alors obtenu la moyenne de l'affaiblissement à donner à la tension de sortie du générateur pour obtenir le seuil au bruit complexe chez le sujet normal. Qu'il s'agisse des fréquences pures ou du bruit complexe, les mesures qui vont suivre ne concernent donc pas la valeur absolue des seuils d'audibilité, mais leur valeur relative vis à vis du sujet normal théorique moyen. Elles vont exprimer, en décibels, les différences existant entre les seuils de ce sujet moyen et ceux des sujets d'expérience. Elles traduiront donc des déficits, soit positifs chez les mieux entendants, soit négatifs chez les sujets normaux moins favorisés ou chez les sujets sourds.

Les épreuves proprement dites ont porté sur 45 oreilles normales et 88 oreilles sourdes, lesquelles ont été examinées parallèlement et avec le bruit complexe 400 ··· 1900 Hz et avec les fréquences pures 500, 800, 1000 et 2000 Hz.

4. Résultats

Chez les sujets normaux, dans 40 cas sur 45, le déficit au bruit complexe 400 ··· 1900 Hz paraît plus grand que celui du son pur le mieux perçu parmi les fréquences 500, 800, 1000 et 2000 Hz. La répartition des écarts ainsi enregistrés est indiquée dans le tableau ci-dessous: 40 fois, ils sont compris entre 0 et 10 dB.

En ce qui concerne les oreilles sourdes, le déficit au bruit complexe est supérieur à celui de la fréquence pure la mieux perçue dans 53 cas sur 88, soit dans près des deux tiers des cas, et il est, au contraire, plus petit dans 29 cas. Dans 6 cas enfin, le déficit est le même. Les écarts sont très habituellement échelonnés entre 0 et 10 dB.

Écarts des déficits pour le bruit complexe 400 ··· 1900 Hz et pour le son pur le mieux perçu parmi les fréquences 500, 800, 1000 et 2000 Hz.

Écarts compris entre:	Sujets normaux		Sujets sourds	
	Son pur mieux perçu	Bruit complexe mieux perçu	Son pur mieux perçu	Bruit complexe mieux perçu
0 et 2 dB	8 cas	2 cas	13 cas	11 cas
3 et 5 dB	18 „	2 „	17 „	7 „
6 et 10 dB	9 „	1 „	12 „	10 „
11 et 15 dB	3 „	0 „	9 „	1 „
16 et 20 dB	1 „	0 „	1 „	0 „
21 et 25 dB	1 „	0 „	1 „	0 „
	40 cas	5 cas	53 cas	29 cas

5. Conclusions

En somme, si l'on voulait totaliser les résultats pour les 45 oreilles normales et les 88 oreilles sourdes, on trouverait que l'audibilité du bruit complexe 400 ··· 1900 Hz s'écarte de celle de la fréquence la

mieux perçue parmi les fréquences 500, 800, 1000 et 2000 Hz dans les proportions ci-après. Sur 133 oreilles, dans 84 cas, l'écart en plus ou en moins, est égal ou inférieur à 5 dB; dans 32 cas, il est compris entre 6 et 10 dB; et 17 fois, il est supérieur à 10 dB.

En ce qui concerne les écarts supérieurs à 10 dB, il convient de spécifier que les sujets examinés, sains ou malades, n'avaient aucune expérience des investigations audiométriques, lesquelles exigent une collaboration très étroite des individus soumis aux mesures, une bonne compréhension du but à atteindre et une attention de tous les instants, qualités parfois difficilement requérables de sujets pris au hasard et non habitués à l'écoute.

Quoiqu'il en soit, les essais que nous avons conduits jusqu'ici n'ont pas actuellement confirmé l'hypothèse de travail, au moins dans les conditions de l'expérimentation: le son pur le mieux perçu a semblé l'être généralement mieux encore que le bruit complexe.

Notons cependant que, si l'on n'opère pas dans un silence rigoureux, ce n'est pas le seuil de la bande 400 ··· 1900 Hz que l'on enregistre, mais vraisemblablement une variation d'intensité des bruits perçus, le bruit de fond et le bruit complexe utilisé cumulant leurs effets. En raison même de la composition de la bande de fréquences utilisée, l'effet de masque doit jouer, dans ce cas, au maximum. Cette possibilité expliquerait pourquoi les sujets sains, sauf exception, ont mieux répondu aux fréquences pures qu'au bruit complexe. Ils auraient été facilement assourdis par un léger bruit de fond, bruit que les sujets malades n'auraient pas perçu ou auraient moins perçu en raison même de leur surdité.

Par ailleurs, en regard de la largeur de la bande complexe, le nombre des fréquences pures explorées a été forcément limité et, dans l'intervalle de celles-ci, il n'est pas exclus que d'autres fréquences pures non examinées aient eu un niveau meilleur. Cette remarque empêche de donner leur véritable signification aux cas dans lesquels le bruit complexe nous a paru mieux perçu que le son pur.

C'est pourquoi nous poursuivons notre expérimentation sur une bande de fréquences plus étroite. Le problème, en effet, n'est pas dénué d'intérêt, tant sur le plan médical, où il est susceptible d'apporter une technique nouvelle d'examen, que sur le plan physique où son étude peut conduire, en particulier, à une définition stricte de l'intensité des bruits complexes à spectre continu, question restant à approfondir.

LA NORMALISATION DES APPAREILS DE PROTHÈSE AUDITIVE EN FRANCE

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Centre National d'Études des Télécommunications, Paris

Sommaire. Depuis la fin de la deuxième guerre mondiale, la construction des appareils de prothèse auditive a fait, dans le monde, des progrès considérables; il était donc normal que le problème de l'étalonnage de tels appareils se posât et que, afin d'éviter la vente des appareils de mauvaise qualité, soient normalisées certaines caractéristiques physiques et mécaniques. En France, depuis Juin 1952, outre les dimensions des piles et des fiches, ont été définies avec précision: des clauses de résistance aux chocs, à la chaleur et à l'humidité; une clause relative à la limite inférieure du gain acoustique; une clause définissant la courbe de réponse des appareils; une clause relative à la distorsion harmonique. Le bruit de fond de l'appareil est également mesuré à titre indicatif.

Zusammenfassung. Bei der Konstruktion von Hörgeräten sind seit Kriegsende beträchtliche Fortschritte gemacht worden und man hat sich nun mit der Eichung und der Normierung gewisser charakteristischer Eigenschaften der Geräte zu befassen. Außer für Batterien und Stecker wurden in Frankreich seit Juni 1952 genaue Definitionen geschaffen für Widerstandsfähigkeit gegen Stoß, Wärme und Feuchtigkeit, Mindestverstärkung, Bedingungen für die Frequenzkurve und die harmonischen Verzerrungen. Auch hinsichtlich des Rauschens wurden Messungen in dieser Richtung ausgeführt.

Summary. Since 1945 considerable progress has been made in apparatus for hearing aids, which has brought with it the problem of standardising such equipment and avoiding the sale of inferior makes. In France since June 1952, beside dimensions of batteries and plugs, standards have been laid down concerning: resistance to vibration, heat and humidity; minimum gain; definition of response curve; harmonic distortion. The background threshold of an aid is also measured.

A. Généralités

Au cours de la dernière guerre, les pièces détachées miniatures furent étudiées à des fins militaires. Il était donc normal, à la fin des hostilités, de chercher leurs applications vers la réalisation d'appareils civils des plus variés. C'est, en particulier, à partir de 1945 qu'apparurent les premiers appareils de prothèse à

amplification électronique; aussi, dès cette époque, commença à se poser le problème de la vérification et par suite de la normalisation des méthodes de mesures de ces appareils, ainsi que celui de la définition des caractéristiques acoustiques à imposer à ce matériel pour la correction des déficiences auditives.

L'objet du présent travail n'est pas de préciser les méthodes de mesures relatives à ces appareils, car des efforts dans ce sens ont déjà été tentés à plusieurs reprises [1], [2], mais, au contraire, de définir les normes qui sont actuellement appliquées en France.

B. Les bases techniques de la normalisation des appareils de prothèse auditive

Lorsqu'il s'agit de normaliser des appareils électroacoustiques, les deux premiers critères auxquels on songe immédiatement sont la caractéristique de fréquence et la distorsion harmonique. Dans le cas des microphones, haut-parleurs ou amplificateurs destinés à des usages classiques, il n'y a pas de difficulté majeure à définir les divers paramètres habituels; par contre, dans le cas des appareils de prothèse auditive, le problème est beaucoup plus délicat, principalement en ce qui concerne la courbe de réponse, car il est jusqu'ici très malaisé de définir, a priori, la correction la mieux adaptée à tel ou tel sourd et de définir la meilleure courbe convenant à la majorité des cas. Tous les spécialistes s'accordent pour admettre, actuellement, que les courbes audiométriques habituelles ne suffisent pas à donner les éléments qui permettent de déterminer la courbe de réponse optima d'un appareil de prothèse, et, comme d'autre part, les courbes d'égale sensation sonore ne sont pas parallèles entre elles chez un sujet normal, il n'y a aucune raison, a priori, pour qu'il n'en soit pas de même chez un sujet sourd. Pour résoudre ce problème, il faudrait disposer d'un système microphone—amplificateur—haut-parleur de haute qualité dont il serait possible, à volonté, de modifier la courbe de réponse, non pas en affaiblissant graduellement les basses ou les hautes fréquences, mais en augmentant ou en diminuant strictement le gain dans des bandes étroites, ayant par exemple une largeur de $\frac{1}{2}$ ou de $\frac{1}{3}$ d'octave. Un tel appareil est actuellement à l'étude dans notre centre de recherches et, lorqu'il sera terminé, nous nous proposons d'entreprendre une telle étude en liaison avec des médecins spécialistes.

Cependant, dès 1949, le Ministère de la Santé Publique, en France, s'inquiétait du grand nombre de types d'appareils vendus et chargea notre organisme de préparer un projet de normalisation le plus rapidement possible. Il ne nous était pas possible à cette époque, pour diverses raisons, d'entreprendre utilement une étude complète sur ce sujet; aussi, convinmes-nous d'adopter comme base de départ pour notre travail les études anglaise [3] et américaine [4] que nous avons complétées par quelques essais personnels.

De son côté, le comité britannique [3] a proposé la courbe représentée par la figure 1, qui est limitée aux fréquences 200 et 4000 Hz; cette courbe croissait de 10 dB par octave de 200 à 800 Hz, puis, à partir de cette limite, offrait deux possibilités:

— soit tracé horizontal (gain constant) de 800 à 4000 Hz;

— soit pente uniforme (gain croissant) de 5 dB par octave.

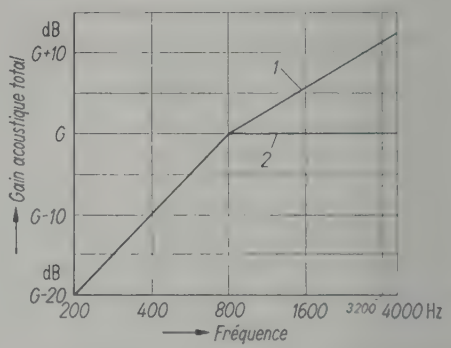


Fig. 1. Caractéristiques proposées par le Comité Britannique. G est le gain acoustique maximum à la fréquence 800 Hz; il doit être supérieur à 40 dB.

Les tolérances admises pour des raisons pratiques étaient de l'ordre de $\pm$ 5 dB de part et d'autre de ces deux courbes. Le gain acoustique, à 800 Hz, devait être au moins égal à 40 dB. Il était également recommandé que la pression de sortie soit proportionnelle à la pression sonore agissant sur le microphone de l'appareil jusqu'à des valeurs de la pression acoustique de 2 newton/m^2 produite par l'écouteur sur l'oreille artificielle; cet essai de linéarité devait être exécuté à la fréquence 750 Hz. Une recommandation était également faite en ce qui concerne le bruit de fond propre de l'appareil. Enfin, des précisions étaient fournies relativement à la diminution du gain acoustique en fonction de la tension de la pile de chauffage: le gain ne devait pas varier de plus de 6 dB lorsque la tension de la pile de chauffage baissait de 1,5 volt à 1 volt. Diverses recommandations relatives aux pièces détachées et à la construction des appareils étaient également formulées, recommandations que la commission française n'a pas accepté de prendre à son compte.

Le comité américain [4], de son côté, propose une courbe de réponse constante de 300 à 4000 Hz environ et recommande des coupures assez brusques au-dessous de 300 Hz et au-dessus de 4000 Hz. Il préconise également l'établissement de deux posi-

tions de tonalité, l'une correspondant à une croissance de la courbe de réponse de 4 à 5 dB par octave, l'autre à une croissance maximum de 9 dB par octave. La figure 2 précise la forme de ces diverses courbes. Ce comité recommande aussi la limitation des niveaux acoustiques produits par les appareils à 4 pressions sonores différentes, selon le type de surdité: 114, 120, 126 et 132 dB audessus de $2 \cdot 10^{-5}$ N/m^2; il préconise la limitation du bruit de fond propre de l'appareil et considère qu'un gain acoustique de 80 dB peut être pris comme un maximum; par contre, la limite inférieure du gain est fixée à 30 dB; enfin, le potentiomètre de réglage du gain doit permettre des variations au moins égales à 40 dB.

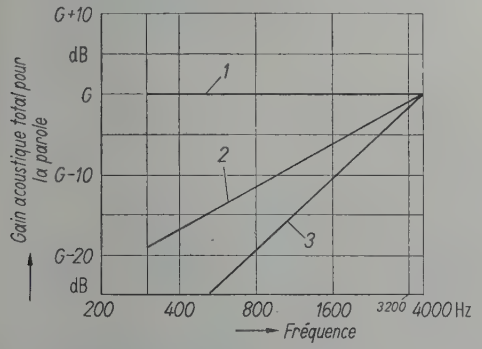


Fig. 2. Caractéristiques proposées par le Comité Américain. *G* est le gain acoustique maximum; il doit toujours être supérieur à 30 dB.

Le comité français attachait une grande importance à ce qu'une normalisation soit faite le plus rapidement possible, afin que la Sécurité Sociale ne rembourse à l'assuré que les appareils homologués; aussi, il fut décidé d'exécuter, en France, sur les appareils existant, les mesures selon les méthodes que nous avions déjà proposées [2] en utilisant l'oreille artificielle qui avait été étudiée au Laboratoire d'Acoustique du C.N.E.T. et qui est représentée sur la figure 3.

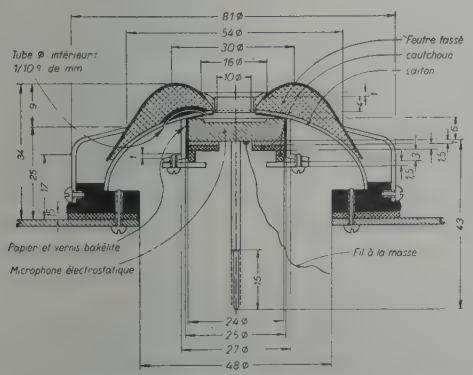


Fig. 3. L'oreille artificielle du Département Acoustique du Centre National d'Études des Télécommunications; cotes en mm.

C. La normalisation des appareils de prothèse en France

En dehors des considérations d'ordre purement acoustique qui viennent d'être rappelées, la commission technique chargée d'établir les normes relatives aux appareils de prothèse, avait décidé de préciser:

— des clauses relatives aux dimensions des fiches de branchement du cordon à l'appareil et à l'écouteur;

— des clauses relatives aux dimensions des piles;

— des clauses de résistance mécanique du boîtier et de l'appareil complet;

— des clauses de résistance à la chaleur et à l'humidité de l'appareil complet.

1. Clauses acoustiques

Il fut décidé que ces clauses ne s'appliqueraient, actuellement, qu'aux appareils à conduction aérienne, le problème de l'étalonnage des vibrateurs à conduction osseuse ne semblant pas, en effet, encore résolu de façon suffisamment précise malgré les derniers travaux, tant objectifs [5] que subjectifs [6], exécutés sur les aspects physiques et physiologiques qu'il présente.

a) amplification (gain acoustique): elle doit être supérieure à 44 dB à la fréquence 1000 Hz.

b) courbe de réponse: elle doit être comprise à l'intérieur du parallélogramme hachuré défini par la figure 4, limité d'une part, par les deux ordonnées d'abscisses 250 et 4000 Hz, et, d'autre part, par deux droites tracées parallélement à une troisième droite prise comme courbe type et présentant une pente positive de 2,5 dB par octave; ces deux droites limites sont distantes de + et — 6 dB de la courbe type. Cependant, entre 3500 et 4000 Hz, la tolérance est portée de ± 6 dB à ± 10 dB.

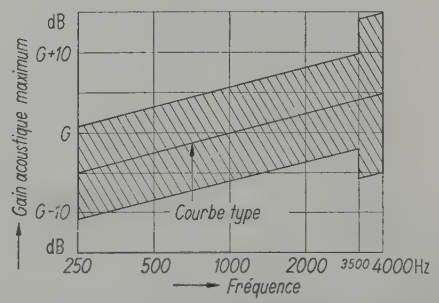


Fig. 4. Caractéristique normalisée en France depuis le 1er janvier 1953. *G* est le gain acoustique mesuré à la fréquence 1000 Hz; il doit être supérieur à 44 dB.

c) distorsion harmonique: elle doit être inférieure à 10% à la fréquence 1000 Hz, pour le gain maximum de l'appareil et pour un niveau

à l'émission de 50 dB au-dessus de $2 \cdot 10^{-5}$ N/m². Toutes ces mesures se font en champ acoustique libre, pour un niveau à l'émission de 50 dB au-dessus de $2 \cdot 10^{-5}$ N/m², à l'aide de l'oreille artificielle déjà représentée sur la figure 3.

2. Fiches de branchement

Au cours de l'examen des appareils régulièrement construits, une vingtaine de modèles différents avaient pu être dénombrés, en ce qui concerne les fiches de connexion des cordons de l'appareil avec

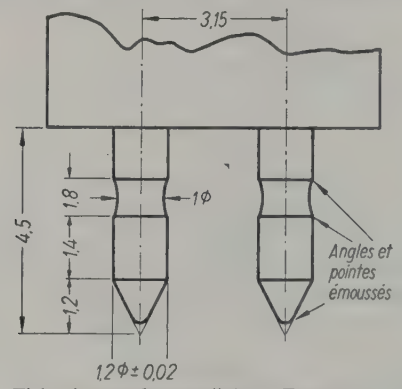


Fig. 5. Fiche du type 1 normalisée en France; cotes en mm.

l'écouteur. Actuellement, deux modèles sont homologués, représentés par les figures 5 et 6, l'un de type américain courant, l'autre de type français. De plus, ces fiches doivent être symétriques, c'est-à-dire identiques du côté appareil et du côté écouteur.

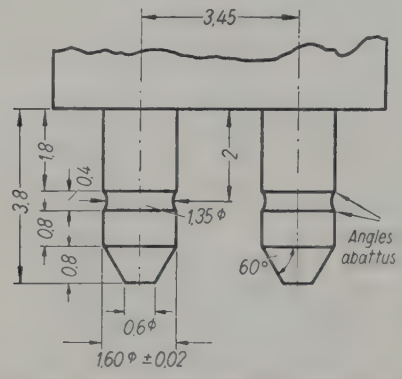


Fig. 6. Fiche du type 2 normalisée en France; cotes en mm.

3. Clauses relatives aux piles

Devant la multiplicité des modèles de piles qui étaient utilisées en 1949 et 1950, ont été homologuées, du point de vue dimensionnel, 6 modèles de piles haute tension (3 modèles miniatures et 3 modèles subminiatures) et 2 modèles de piles (ou accumulateurs) basse tension. Le tableau met en évidence les dimensions et les tolérances acceptées; les cotes ne comportant aucune tolérance sont des valeurs maxima.

Tableau

Piles haute tension

Tension V	Épaisseur mm	Largeur mm	Hauteur mm
15	16	16	35 ± 0,5
	26	16	37 ± 0,5
22,5	16	16	50 ± 0,5
	26	16	50 ± 0,5
30	32	16	37 ± 0,5
	26	16	63 ± 0,5

Piles et accumulateurs basse tension

Tension V	Diamètre mm	Hauteur mm
1,25	15	50 ± 0,5
1,5	15	50 ± 0,5
1,5	21,5	$61{,}5\ ^{+\,0}_{-\,0{,}5}$

4. Clauses de résistance mécanique

Ces clauses se divisent en deux catégories, l'une relative à des essais de résistance aux chocs, l'autre à des essais par "secousses".

a) essai de chocs: cet essai consiste à faire tomber sur le boîtier de l'appareil une bille de bois de 150 grammes, d'une hauteur de 5 cm, avec une vitesse initiale nulle. On fait exécuter à cette bille 10 chutes successives en 10 points différents du boîtier, qui ne doit présenter, après l'épreuve, aucun bris ni aucune cassure.

b) essai de "secousses": l'appareil, en ordre de marche, solidement fixé sur un support pesant 2 kg, subit durant 8 heures consécutives des chutes d'une hauteur de 2 cm à la cadence de 4 par minute (soit près de 2000 secousses). Après cet essai, l'appareil doit toujours satisfaire aux clauses acoustiques.

5. Clauses de résistance à la chaleur et à l'humidité

L'appareil complet, non muni de ses piles, est placé durant 8 heures dans une enceinte portée à une température de 30° ± 2° où l'atmosphère a un degré hygrométrique compris entre 80 et 90%. Après cet essai, l'appareil doit toujours satisfaire aux clauses acoustiques.

D. Conclusions

Les diverses clauses et spécifications dont l'origine technique et la nature ont été précisées dans le présent travail ont fait l'objet d'un arrêté paru au Journal Officiel de la République Française [7] en juin 1952 et qui est entré en application le 1er

janvier 1953; depuis cette date, une vingtaine d'appareils de modèles différents ont été présentés en vue de leur homologation. Sur ce nombre, une dizaine seulement a été agréée à titre définitif, et cinq ou six à titre provisoire; l'agrément pouvant, d'ailleurs, toujours être retiré après examen exécuté sur les séries de fabrication.

Il est évident qu'une norme adoptée en une matière aussi complexe, qui fait heureusement habituellement l'objet d'études systématiques, ne peut prétendre à être parfaite. C'est le sort de toute norme que de susciter des critiques plus ou moins justifiées; du moins, elle a eu aussi le mérite d'améliorer dès maintenant la fabrication électrique et mécanique des appareils ainsi que l'a confirmé l'examen de systèmes récents et d'encourager certains industriels français à entreprendre l'étude de nouveaux modèles. Son objet était d'éliminer le matériel de qualité nettement insuffisante; elle n'est ni immuable ni définitive et pourra, dans l'avenir, être modifiée si l'expérimentation pratiquée rationnellement en fait apparaître la nécessité. Elle a, cependant, l'avantage positif d'exister et de pouvoir, ainsi, guider les constructeurs dans leurs réalisations. Ces derniers conservent naturellement toute liberté d'inclure, s'ils le désirent, dans leurs appareils plusieurs positions de tonalité ou plusieurs possibilités de filtrage. Toutefois, une au moins des courbes de réponse doit correspondre à la clause acoustique imposée; de même, le contrôle de volume automatique n'a pas été normalisé, les informations précises à ce sujet ne semblant pas encore suffisantes pour qu'une spécification puisse être formulée à cet égard. Ce travail sera l'objet de futures recherches au cour des années à venir.

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THE NORMAL THRESHOLD OF HEARING AND OTHER ASPECTS OF STANDARDISATION IN AUDIOMETRY

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Summary. The paper describes recent work at the National Physical Laboratory on audiometric standards, with particular reference to the determination of the normal threshold of hearing and developments in the calibration of bone-conduction receivers. The N.P.L. threshold measurements agree closely with the results of a later investigation by the Central Medical Establishment of the R.A.F. and are in fair agreement with those of Sivian and White. The mechanical impedance of the human mastoid process has been measured, and the results applied to the design of an "artificial mastoid" for the objective calibration of bone-conduction receivers.

Sommaire. On indique dans cet article les travaux récents effectués au National Physical Laboratory sur les étalons audiométriques, et, en particulier, sur la détermination du seuil normal d'audition et sur l'étalonnage des récepteurs par conduction osseuse. Les résultats des mesures du seuil faites au N.P.L. sont en excellent accord avec ceux d'une étude ultérieure entreprise par le Central Medical Establishment de la R.A.F., et cadrent bien avec ceux de Sivian et White. On a mesuré l'impédance mécanique de la mastoïde humaine, et les résultats obtenus ont servi à établir une "mastoïde artificielle" destinée à l'étalonnage objectif des récepteurs par conduction osseuse.

Zusammenfassung. Es werden neuere im National Physical Laboratory durchgeführte Arbeiten über audiometrische Normen beschrieben, wobei besonders die Bestimmung der normalen Hörschwelle und Entwicklungen auf dem Gebiet der Eichung von Knochenleitungshörern behandelt werden. Die Messungen der Hörschwelle im N.P.L. stimmen eng überein mit den Ergebnissen einer späteren Untersuchung des Central Medical Establishment der R.A.F. und sind ebenfalls in ziemlich guter Übereinstimmung mit denen von Sivian und White. Die mechanische Impedanz des menschlichen Schläfenbeinfortsatzes wurde gemessen und die Ergebnisse wurden für die Konstruktion eines „künstlichen Fortsatzes" für die objektive Eichung von Knochenleitungshörern verwendet.

1. Introduction

The need for a standard reference zero for audiometry, corresponding with the normal threshold of persons of unimpaired hearing, has been apparent for some time, and the problem has been recognised at a number of recent international conferences [1], [2], [3]. In Great Britain consideration of this problem received a strong impetus in 1939 when, at the instance of the Royal Society of Medicine, the British Standards Institution commenced work, which was interrupted by the war, on the preparation of performance standards for pure-tone audiometers. Further initiative was taken by the Medical Research Council, through the medium of its Electro-Acoustics Committee which was concerned with the improvement of audiometer standards concurrently with its work on the specification of the national hearing aid. At the instance of these bodies a programme of experimental work was undertaken at the National Physical Laboratory, in the course of which a thorough investigation was made of the threshold of hearing by earphone listening on a group of about 100 otologically normal subjects in the age group 18 ··· 25 years in accordance with the recommendation of the Electro-Acoustics Committee [4]. It is the main object of this paper to summarise the present position of these investigations and to indicate the further steps towards the establishment of standards which are now being taken, or are envisaged.

2. The normal threshold of hearing

The first discussion of note on the threshold of hearing appeared in the classic paper by SIVIAN and WHITE [5] in which the authors reviewed the very conflicting data available in 1933, and produced a curve of threshold pressure in dependence on frequency which represented their assessment of these results. Since that time two surveys on large groups of subjects have been carried out in the U.S.A. which gave threshold levels of the order 10 to 15 dB higher than SIVIAN and WHITE's values [6]. To judge from the discussion of these surveys given by STEINBERG, MONTGOMERY and GARDNER [7], [8] it does not appear that any definite conclusion can be drawn regarding the origin of these discrepancies.

The investigation recently carried out at the National Physical Laboratory on a group of 100 otologically normal subjects was shortly followed by work at the Central Medical Establishment of the Royal Air Force in which over 500 normal subjects in the same age group were examined. These investigations, which have been described in detail elsewhere [9], [10], gave very closely concordant results, the discrepancy at any one frequency being not greater than 1.5 dB, and the average discrepancy over the frequency range 250 ··· 8000 c/s not greater than 0.5 dB. This is quite remarkable agreement, and in the view of the author provides strong evidence for the reliability and representative nature of the results. It is interesting to note that these measurements are in very fair agreement with the curve suggested in 1933 by SIVIAN and WHITE, over the range of frequency (250 ··· 8000 c/s) of most importance in audiometry, being within $\pm$ 5 dB at the various frequencies in this range with an average discrepancy of the order of only 0.5 dB. They diverge from SIVIAN and WHITE's values in the direction of increased sensitivity, however, at the very low and very high frequencies.

The combined results of the investigations at the National Physical Laboratory and at the Central Medical Establishment of the Royal Air Force, covering a total of over 1200 ears, are given in Fig. 1 and Table I. The values in Fig. 1 are the normal minimum audible pressures measured at the entrance to the external auditory meatus, and are shown in comparison with the results of the other investigations mentioned above. In Table I are given the equivalent threshold sound pressures set up by an earphone of a particular pattern when coupled to an artificial ear of the type employed at the National Physical Laboratory and described in reference 10. The type of earphone in question is that which is now maintained at the National Physical Laboratory as a reference standard for the purpose of audiometer calibration, but the corresponding figures for other patterns of earphone may easily be derived, e.g. by the process of equal-loudness balancing.

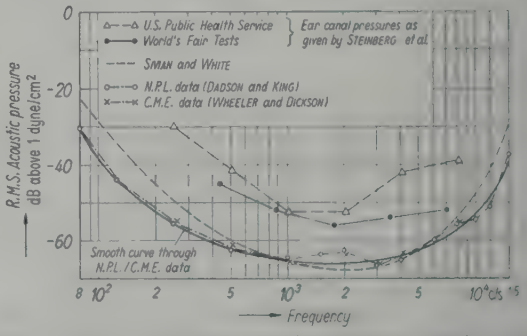


Fig. 1. N.P.L. and C.M.E. data for minimum audible pressure at entrance to external auditory meatus, compared with other data.

The "equivalent" threshold pressures (Table I) are those set up in the artifical ear with the same voltage at the earphone as would produce the minimum audible pressure in human ears (Fig. 1).

The combined N.P.L. and C.M.E. results given in Fig. 1 and Table I probably constitute the most coherent and representative data available at the present day. They are considered to be accurate to

Table I: Normal equivalent threshold sound pressures in artificial ear of N.P.L. type for 4026A pattern earphone corresponding to the results in Fig. 1.

Frequency c/s	Normal equivalent threshold sound pressure in N.P.L. artifical ear for 4026A pattern earphone (dB above 1 dyne/cm² *)
80	— 13.5
125	— 29.5
250	— 44.5
500	— 62
1000	— 68
1500	— 66 5
2000	— 65
3000	— 68
4000	— 65
6000	— 65
8000	— 65
10000	— 58.5

the order of 1 dB under the stated conditions of test, and are the present basis of all audiometer calibration work carried out at the National Physical Laboratory. The values are under consideration with a view to their issue as a British standard, and the responsible British Committee has put them forward for consideration by the appropriate committee of the International Standards Organisation as the basis for an internationally agreed reference level. It is hoped that these steps will go some way to improve the overall accuracy and consistency of audiometric standards and measurements.

3. Bone conduction

We have so far considered only the standardisation of the level of *airborne* sound supplied by the earphone of an audiometer, but the measurement of bone-conducted sound presents problems of a similar nature and on which very little evidence is so far available. The calibration of a bone-conduction receiver is in essentials quite parallel to the calibration of an earphone, and thus as regards objective measurements on bone conduction receivers the first requirement is an "artificial mastoid" which will present to the receiver the same mechanical impedance as the normal human mastoid process. Before such an instrument can be devised, measurements of the mechanical impedance of the human mastoid process are required. An investigation of the mechanical impedance of the mastoid processes of 20 subjects has recently been carried out at the National Physical Laboratory, using patterns of bone conduction receiver typical of those employed for audiometers and hearing aids. Some typical results of these measurements are shown in Figs. 2 and 3, which also indicate the form of equivalent network suitable

* 1 dyne/cm² = 0.1 N/m².

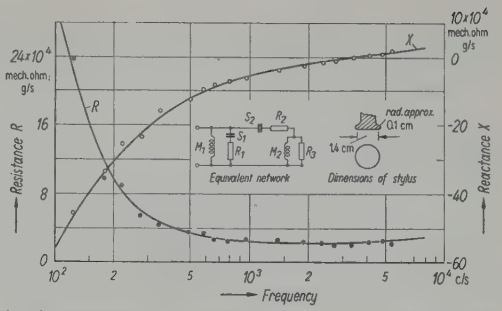


Fig. 2. Mechanical impedance $R + jX$ of mastoid process to "audiometer type" stylus. Force of application 1000 gram weight.

for representation of the results over the frequency range 100 ··· 8000 c/s. The effects of varying the force of application of the receiver and the area of contact with the mastoid process have also been studied. The full results of this work, which gives very comprehensive data on the mechanical impedance of the mastoid process, are in course of publication, in which account is being taken of the few results available from other sources [11], [12], [13]. The results obtained are now being applied to the design and construction of an artificial mastoid for the objective calibration of bone conduction receivers, which is now nearing completion. It is hoped that this work will lead to considerable improvement in a field of measurement in which data has up to the present been almost entirely lacking.

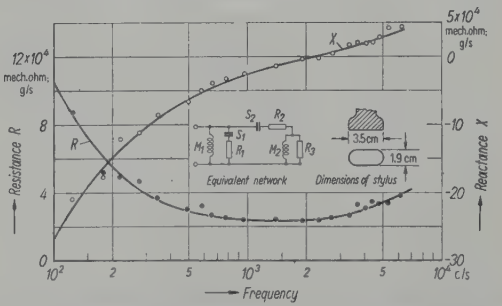


Fig. 3. Mechanical impedance $R + jX$ of mastoid process to "hearing-aid type" stylus. Force of application 400 gram weight.

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Discussion

AYERS, E. W.: To avoid the chaos now existing due to premature standardisation of artificial ears, let us first be sure of agreement on results of mastoid measurements and on their appropriateness.

LECTURER: As a result of the N.P.L.'s work we already have far more information on the mastoid impedance than we ever had on the ear-canal impedance. The data show greater consistency and owing to the relative absence of wave motion effects the impedance curves are simpler in form. Whilst careful discussion of any proposed standards will be necessary there seems no reason for anxiety in this connection.

COOK, R. K.: The average difference between the threshold pressures in the ear canal (as given in the Journal of Laryngology and Otology) for the U.S. Public Health Service data and the National Physical Laboratory data is about 15 dB. However, thanks to co-operation between the N.P.L. and the National Bureau of Standards, this difference was recently redetermined and was found to be slightly less than 10 dB. This now appears to be the best estimate of the difference; the threshold pressures in the ear-canal for the U.S. Public Health Service data of 1936 are, on the average, 10 dB greater than those recently determined at the N.P.L.

LECTURER: One difficulty in interpreting the results of the U.S. Public Health Service Survey is that there seem to be two versions of the ear-canal pressures corresponding to the normal thresholds obtained in that investigation. The results shown in Fig. 1 of the paper were taken from the papers by STEINBERG, MONTGOMERY and GARDNER cited, which as far as we knew were the only estimates of the actual *ear-canal* pressures published. As will be seen from Fig. 1 these results are higher than the British figures for substantially the same age group by amounts varying from 10 to 25 dB (average 18 dB) over the range 250 to 8000 c/s. The intercomparison of earphone standards between N.P.L. and the National Bureau of Standards, mentioned by Dr. COOK, does not actually affect these particular discrepancies. The intercomparison reveals only the differences between the British results and the reference levels adopted at the N.B.S. As Dr. COOK says, the latter reference level differs from the British results by a little less than 10 dB on the average, so it seems that the N.B.S. reference level differs quite considerably from the results published by STEINBERG, MONTGOMERY and GARDNER, although both sets of data are attributed to the same survey. Dr. COOK's statement helps to clarify this position, but it appears that care will still be necessary in discussion of the Public Health Survey investigation to avoid confusion resulting from these different interpretations of the results.

MORTON, J. Y.: A slide was shown giving the mean impedance of 13 normal mastoids which had been measured in 1950. The mean impedance could be represented by a simple series resonant circuit of 0.73 g mass, 0.0075×10^{-6} cm/dyne compliance and 16 000 dyne/(cm/s) resistance over the frequency range 400 to 4000 c/s. This work had been carried out as a preliminary experiment to gain experience before measurements were made on deaf subjects who had been wearing bone conduction receivers, but unfortunately it had not been possible to continue the work until quite recently. Measurements on three subjects who had been wearing receivers for more than a year indicated that they may not form part of the same statistical population as normal subjects. Their compliances were of the order of half, and their resistances of the order of twice the mean of the normal subjects. Although the results for normal subjects would be suitable for an artificial mastoid for audiometer receivers for subjects who had not worn hearing aid receivers, it was possible that they could be improved upon for an artificial mastoid for hearing aid bone conduction receivers.

LECTURER: The reason why Mr. MORTON's results may be represented by a simple series resonant mechanical circuit is the comparatively limited frequency range involved. For accurate representation of the N.P.L. results, which cover the range 100 to 6000 c/s, a seven element network is necessary; this has the additional advantage of throwing far more light on the factors controlling the behaviour of the mechanical system. With regard to possible effects on the mastoid impedance due to wearing a bone conduction receiver, it seems doubtful whether tests on three subjects are sufficiently representative. Further complication seems likely, since the effect, when established, may well depend on the length of time the bone conductor has been worn and on other "user" factors. It seems doubtful whether it would be justifiable to make a serious attempt to take such factors into account in the design of an artificial mastoid, whether for hearing aids or for audiometers. It is in fact to be hoped that, on account of the relatively smooth character of the curves relating mechanical impedance to frequency, the simulation of the mastoid impedance will not be unduly critical.

ÜBER DIE OBERE GRENZFREQUENZ VON HÖRGERÄTEN

von W. Güttner und C. Starke
Siemens-Reiniger-Werke A.G., Erlangen

Zusammenfassung. Um eine natürliche Wiedergabe von Sprache mit den typischen individuellen Feinheiten zu erreichen, ist es das Ziel der Entwicklung hochwertiger Hörhilfen, das übertragene Frequenzband zu erweitern. Maßgeblich für die Beurteilung ist die Frequenzabhängigkeit der akustischen Verstärkung und des maximalen Ausgangspegels. Die meisten Geräte haben bisher eine obere Grenzfrequenz von etwa 2,5 kHz, die sich aus den benutzten Hörern und Ohrpaßstücken ergibt. Es wird gezeigt, daß sich eine Erweiterung bis auf 4,6 kHz dadurch erreichen läßt, daß man die Übertragungseigenschaften der bisherigen akustischen Bauelemente verbessert und zu offenen Ohrpaßstücken übergeht.

Summary. It is the object of the development of high-fidelity reproduction to widen the frequency band of reproduction. To this end the frequency response of the acoustic amplification and the maximum output is important. In the past the upper limit was about 2.5 kc/s, limited by the microphones and earmoulds used but an extension to 4.6 kc/s is possible if one improves the acoustic units in previous constructions and adopts open earmoulds.

Sommaire. Pour améliorer la qualité des appareils d'audition, on cherche à en élargir la bande de fréquence transmise. La courbe de fréquence de l'amplification acoustique et du niveau sonore maximum à la sortie est déterminante à cet égard. La plupart des appareils actuels ont une limite supérieure de fréquence d'environ 2,5 kHz qu'on peut étendre jusqu'à 4,6 kHz en améliorant les caractéristiques de transmission et en utilisant des adaptateurs du type ouvert.

Im allgemeinen soll der Zweck einer Hörhilfe sein, dem Schwerhörigen die Sprache seiner Umwelt verständlich zu machen. Für die Sprachübertragung ist es erforderlich, den normalen Sprachpegel in einem bestimmten Frequenzband auf einen Pegel angenehmen Hörens zu verstärken. In der Fernsprechtechnik kommt man mit einer oberen Frequenzgrenze von etwa 3,5 kHz aus. Dabei ist die Silbenverständlichkeit ca. 90% und ein fortlaufender Satz wird richtig und vollständig gedeutet. Doch ist die Verständlichkeit kein ausreichendes Kriterium für die optimale Übertragungsqualität. Für eine natürliche Wiedergabe von Sprache mit den typischen individuellen Feinheiten ist eine höhere Grenzfrequenz erforderlich. Es muß deshalb das Ziel der Entwicklung hochwertiger Hörhilfen sein, das übertragene Frequenzband zu erweitern.

1. Kriterien für die obere Frequenzgrenze

Die wesentlichen Übertragungseigenschaften einer Hörhilfe lassen sich durch Messung der Frequenzabhängigkeit von akustischer Verstärkung und maximalem Ausgangspegel beurteilen. Die Verstärkung wird bei einem Eingangspegel von etwa 50 dB aufgenommen (der Verstärkungsregler des Hörgerätes ist so eingestellt, daß der lineare Teil der Aussteuerungskennlinie im ganzen Übertragungsbereich nicht überschritten wird). Sie setzt sich aus den Übertragungseigenschaften von Mikrophon, Verstärker und Hörer zusammen. Die meistens benutzten elektromagnetischen Einsteckhörer haben eine Eigenfrequenz von 1 bis 2 kHz; oberhalb dieser Resonanz fällt die Übertragungsgröße ab. Die Kristallmikrophone werden bis zu einer Abstimmung von 4 bis 5 kHz gebaut, wobei im allgemeinen der Anstieg bis zu ihrer Resonanzstelle zur Kompensation der Hörereigenschaften für die Übertragung des Gerätes ausgenützt wird. Es resultiert dann ein Frequenzband, das etwa bis zur Eigenfrequenz des Mikrophons reicht. Die Frequenzabhängigkeit des Ausgangspegels hat dagegen bei der üblichen Anpassung des Hörers an das Endrohr eine weitgehende Ähnlichkeit mit dem bei konstanter elektrischer Leistung erzeugten Hörerschalldruck. Das Endrohr und die Übertragungsgröße des Hörers bestimmen den erreichbaren Dynamikbereich der Hörhilfe.

Bei der Wechselrede zweier Personen rechnet man mit einem Sprachpegel von 50 bis 60 dB (Dynamik $\pm$ 15 dB) bei 1 kHz und etwa 40 bis 50 dB (Dynamik $\pm$ 12 dB) bei 4 kHz. Dieser sollte von der Hörhilfe mit gleicher Dynamik übertragen werden. Fällt aber der maximal erreichbare Ausgangspegel analog der Hörerkurve im oberen Frequenzgebiet stark ab, während sich die Wiedergabekurve bei kleinerem Ausgangsschalldruck im gleichen Frequenzgebiet auf konstantem Verstärkungsniveau befindet, so ist dort der lineare Aussteuerbereich des Gerätes so stark eingeengt, daß die Originaldynamik bereits beschnitten wird. Die obere Grenzfrequenz der Hörhilfe ist also nicht allein von der akustischen Verstärkung sondern auch von ihrem Dynamikbereich abhängig.

2. Frequenzbandbegrenzung durch Hörerankoppelung an das Ohr

Für die weitere Betrachtung wird vorausgesetzt, daß Verstärkung und maximal erreichbarer Ausgangspegel der Hörhilfe für eine optimale Über-

tragung im Gebiet oberhalb 1 kHz annähernd konstant sein sollen, eine Bedingung, die anhand von Verständlichkeitsmessungen mit einer praktisch idealen Übertragungsanlage [1] für die meisten Schwerhörigkeitsfälle als erstrebenswert ermittelt wurde.

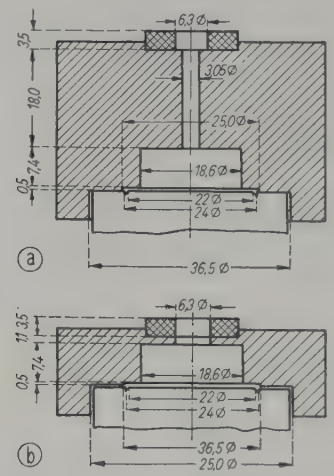


Abb. 1. (a) Typ-2 Druckkammer [2] für herkömmliche Ohrpaßstücke (Maßstab mm);
(b) Modifizierte Typ-2 Druckkammer [3] für „offene" Ohrpaßstücke.

Die obere Grenzfrequenz eines Hörgerätes kann auf Grund der Übertragungseigenschaften der benutzten Hörer beurteilt werden. Es ist üblich, die Hörermessungen an einem schallharten Ohr (Typ-2 Druckkammer, Abb. 1a, [2]) vorzunehmen, da im Gegensatz zu sonstigen künstlichen Ohren hier die Ergebnisse reproduzierbar bleiben. Das 2 cm³ Volumen (18,6 mm ø) entspricht einer mittleren Gehörganggröße bei Abschluß mit einem Einsteckhörer, und der in der Druckkammer befindliche lange Kanal bildet die in den herkömmlichen Ohrpaßstücken (Abb. 2a) benutzten Bohrungen nach.



(a) (b)

Abb. 2. (a) Herkömmliches Ohrpaßstück;
(b) „Offenes" Ohrpaßstück.

Hörer und Druckkammer stellen ein gekoppeltes Schwingungssystem dar, dessen mechanisches Ersatzbild in Abb. 3 gezeigt ist. Die Hörermembran ist durch Masse m_M, Steife s_M und Reibungswiderstand r_M gegeben. Das vor der Membran befindliche Hörervolumen V_V stellt im wesentlichen eine Steife s_V dar, die Bohrung der Druckkammer ist durch eine Masse m_D und einen Widerstand r_D und das 2 cm³ schallharte Druckkammervolumen V_D durch die Steife s_D gegeben, solange alle linearen Dimensionen des Tonraumes klein zur Wellenlänge sind.

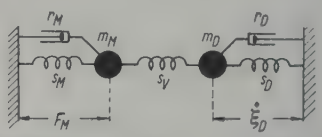


Abb. 3. Mechanisches Ersatzbild des Systems Hörer und Druckkammer.

Der mechanische Scheinwiderstand der Membran ist

$$W_M = r_M + \mathrm{j}\,(\omega m_M - s_M/\omega), \tag{1}$$

der des Hörervolumens

$$W_V = -\mathrm{j}\,s_V/\omega = -\mathrm{j}\,\frac{\varrho c^2}{\omega V_V}\,S_M^2 \tag{2}$$

und der des angeschlossenen Druckkammersystems

$$W_D = r_D + \mathrm{j}\,(\omega m_D - s_D/\omega), \tag{3}$$

worin s_D analog zu (2), und

$$r_{D(L\gg R)} = \frac{L}{\pi R^3}\sqrt{2\mu\varrho\omega}\;S_M^2, \tag{4}$$

$$m_{D(L\gg R)} = \left(\frac{\varrho L}{\pi R^2} + \frac{L}{\omega\pi R^3}\sqrt{2\mu\varrho\omega}\right)S_M^2 \tag{5}$$

einzusetzen sind, und S_M die effektiv strahlende Fläche der Hörermembran, ϱ, c und μ Dichte, Schallgeschwindigkeit und Reibungskoeffizient der Luft und L und R Länge und Radius des Druckkammerkanals sind.

Der Schalldruck p_D in der Druckkammer läßt sich bei einer erregenden Kraft F_M der Membran errechnen:

$$p_D = -\mathrm{j}\,\frac{\varrho c^2}{\omega V_D}\,S_D \cdot \frac{F_M}{W_M\left(1 + \dfrac{W_D}{W_V}\right) + W_D}. \tag{6}$$

S_D ist die effektiv strahlende Fläche des Druckkammerkanals.

Unter Vernachlässigung der Dämpfungen berechnen sich zwei Eigenfrequenzen:

$$\omega_{1,2}^2\,(m_M + m_D) - (s_M + s_D) - \\ - \frac{1}{s_V}\,(\omega_{1,2}^2\,m_M - s_M)\,(\omega_{1,2}^2\,m_D - s_D) = 0. \tag{7}$$

Für die obere Grenzfrequenz des zu übertragenden Bandes ist die Lage der Eigenfrequenz ω_2 maßgebend. Darüber fällt der Schalldruck mit ω^4 ab, d.h. der im Schallweg liegende gekoppelte Tonraum (W_V und W_D) wirkt zusätzlich als Tiefpaß und drückt den oberhalb der Membranresonanz abfallenden Schalldruck noch weiter herab.

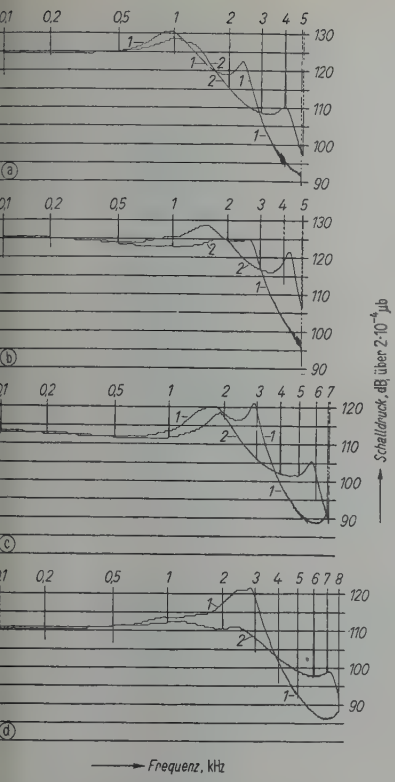


Abb. 4. Hörerschalldruck bei etwa konstanter Leistung von 1 mW im Nennwiderstand 50 Ω (Anschluß des Hörers an eine Spannungsquelle EMK = 1,12 V und $R_i = 200\,\Omega$, Anpassung etwa die gleiche wie am Ausgang der Hörhilfe):
(a) Membrandurchmesser ca. 18 mm;
(b) dgl. mit angekoppeltem Helmholtzresonator;
(c) Membrandurchmesser ca. 15 mm;
(d) Membrandurchmesser ca. 11 mm.
Kurve 1: Typ-2 Druckkammer (Abb. 1a);
Kurve 2: Mod. Typ-2 Druckkammer (Abb. 1b).

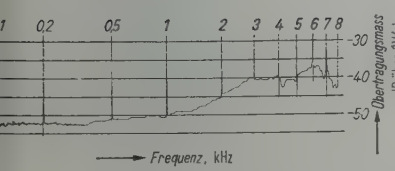


Abb. 6. Übertragungsmaß des Kristallmikrophons.

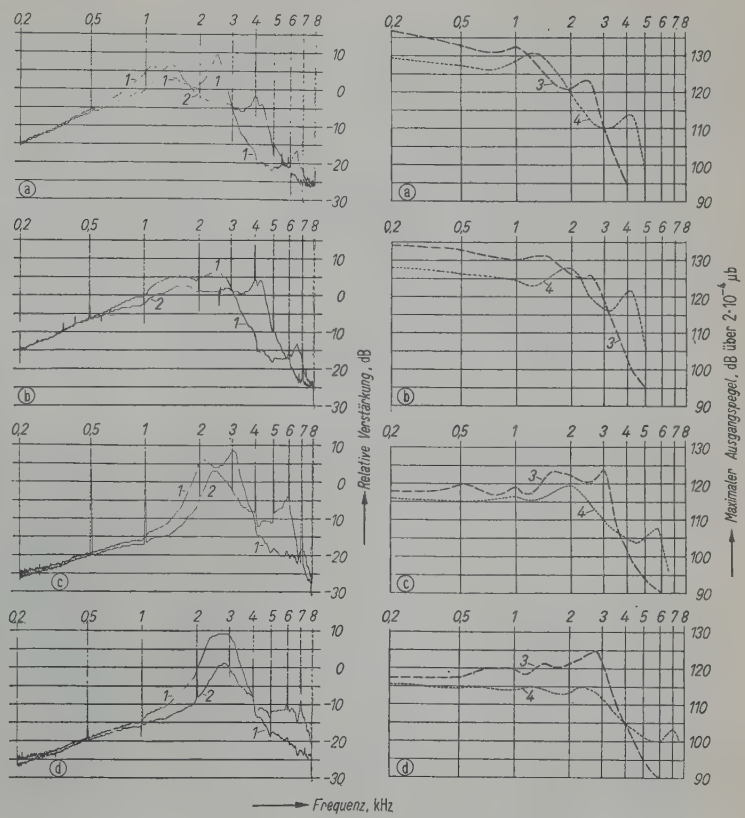


Abb. 5. Akustische Verstärkung und maximaler Ausgangspegel einer Hörhilfe mit den Hörern der Abb. 4 a, b, c, d analog Abb. 4.
Kurve 1: Verstärkung
Kurve 3: max. Ausgangspegel } Typ-2 Druckkammer (Abb. 1a)
Kurve 2: Verstärkung
Kurve 4: max. Ausgangspegel } Mod. Typ-2 Druckkammer (Abb. 1b).

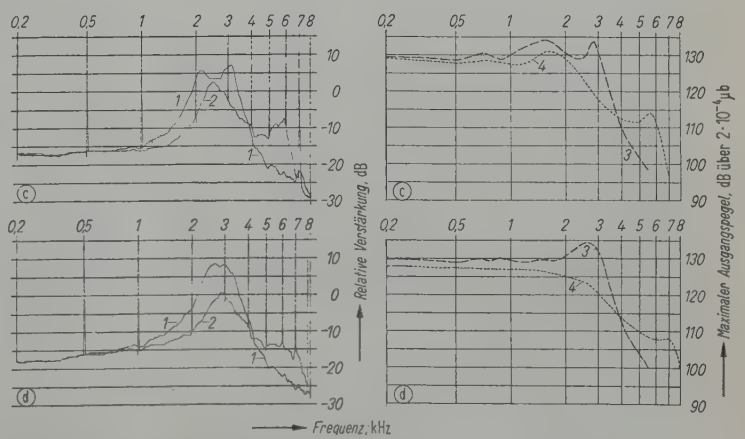


Abb. 7. Akustische Verstärkung und max. Ausgangspegel einer Hörhilfe mit Gegentaktendstufe und den Hörern der Abb. 4c und d;
c und d, Kurve 1 bis 4 analog Abb. 4 bzw. 5.

Mit den üblichen Hörer- und Druckkammerabmessungen (Abb. 1a) ergibt die Rechnung für die obere Resonanzstelle einen Wert von 2,5 kHz. Durch Verkleinerung des vor der Membran liegenden Volumens V_V läßt sich dieser Wert noch erhöhen. Aus technischen Gründen kann jedoch eine bestimmte Größe von V_V nicht unterschritten werden. Benützt man dagegen ein „offenes" Ohrpaßstück (Abb. 2b), so läßt sich die zweite Resonanzstelle des gekoppelten Schwingungssystems durch Verkleinerung der Masse m_D heraufsetzen. Für die objektive Messung der Übertragungseigenschaften mit einem solchen Ohrpaßstück ist eine modifizierte Typ-2 Druckkammer (Abb. 1b) [3] festgelegt worden, bei der das Verbindungsrohr zwischen Hörer und Druckkammervolumen entsprechend dem benutzten Ohrpaßstück gestaltet werden kann.

3. Experimentelle Ergebnisse

Wie aus Abschnitt 1 hervorgeht, wird die obere Grenzfrequenz einer Hörhilfe entscheidend durch die Frequenzabhängigkeit der Übertragungsgröße des Hörers beeinflußt. In Abb. 4 sind die Übertragungseigenschaften einiger Hörer zusammengestellt. Abb. 4a zeigt einen Hörer üblicher Bauart. Die zwei Resonanzstellen des Hörer-Druckkammersystems für das normale Ohrpaßstück liegen bei 1 und 2,5 kHz (Kurve 1), wie auch die Rechnung ergibt. Mit einer Abweichung von ca. $\pm$ 5 dB ist die Übertragungskurve bis etwas über 2,5 kHz konstant. Benutzt man für den Hörer das „offene" Ohrpaßstück — modifizierte Typ-2 Druckkammer —, so wird zwar die zweite Resonanzstelle auf 4 kHz hinaufgeschoben (Kurve 2), jedoch fällt zwischen 2 und 3 kHz der Schalldruck wesentlich schneller ab als bei der ersten Messung. Anders liegen die Verhältnisse bei dem Hörer der Abb. 4b. Hier wurde zusätzlich an die Hörermembran ein gedämpfter Helmholtzresonator angekoppelt. Mit üblichem Ohrpaßstück reicht das Band bei $\pm$ 5 dB Toleranz bis 3 kHz (Kurve 1), mit „offenem" Ohrpaßstück dagegen bis 4,6 kHz (Kurve 2).

Eine Verkleinerung des Hörersystems und damit der Membran, die dadurch eine größere Steife und höhere Eigenresonanz erhält, sollte eine Verbreiterung des Frequenzbandes bewirken. Das ist jedoch nur bedingt der Fall. In Abb. 4c und d sind die Übertragungseigenschaften kleinerer Hörer dargestellt. In beiden Fällen liegt bei Benutzung des normalen Ohrpaßstückes die obere Grenzfrequenz bei 3,3 kHz. Bei Verwendung des „offenen" Ohrpaßstückes ist zwar die zweite Resonanzstelle auf ca. 6 kHz hinaufgeschoben, jedoch liegt bei dem Hörer der Abb. 4d wegen des Abfalls im oberen Gebiet die Grenzfrequenz bei 4 kHz, bei dem der Abb. 4c sogar schon bei etwa 3 kHz. Außerdem ist der Absolutwert des Schalldruckes um 12 bis 14 dB niedriger als bei den Hörern üblicher Größe.

In Abb. 5 sind Messungen an einer Hörhilfe mit den Hörern der Abb. 4 gezeigt. Das Übertragungsmaß des Mikrophons ist in Abb. 6 wiedergegeben. Die Verstärkungskurven wurden bei gleicher Stellung des Lautstärkereglers gemessen, die Kurven maximalen Ausgangspegels bei voller Verstärkung der Hörhilfe.

Die Messungen bestätigen, daß der Hörer mit angekoppeltem Helmholtzresonator und „offenem" Ohrpaßstück die am besten ausgeglichene Verstärkungskurve liefert (Abb. 5b), wie schon aus den Hörerkurven hervorging. Die obere Grenzfrequenz liegt bei 4,5 kHz und unter Berücksichtigung des maximalen Ausgangspegels kann die Hörhilfe bei Zugrundelegen des normalen Sprachpegels bis zu einer Verstärkung von 50 dB bei 1 kHz ohne Einengung der Dynamik betrieben werden.

Bei der Benutzung der kleineren Hörer zeigen die Verstärkungskurven größere Resonanzüberhöhungen (Abb. 5c und d), und aus dem maximalen Ausgangspegel geht die geringe Aussteuerbarkeit hervor. Um den kleineren Wert der Übertragungsgröße dieses Hörers zu kompensieren, muß die Hörhilfe mit einer leistungsfähigeren Endstufe versehen werden. Abb. 7c und d zeigt die analogen Kurven wie Abb. 5c und d nach Einbau einer Gegentaktendstufe. Die Frequenzabhängigkeit von Verstärkung und maximalem Ausgangspegel ist natürlich die gleiche wie bei der einfachen Endstufe, der Absolutwert der Aussteuerungsgrenze liegt jetzt aber wie bei den Hörern normaler Größe ebenfalls bei ca. 130 dB.

Schrifttum

[1] Davis, H., usw. Hearing aids. Harvard University Press, Cambridge, Mass. 1947.
[2] American standard method for the coupler calibration of earphones. Z 24. 9—1949.
[3] American standard method for measurement of characteristics of hearing aids. Z 24. 14—1953, Fig 4.

ARTIFICIAL EAR DESIGN CRITERIA

by W. W. LANG *)
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Summary. The performances of four earphone calibrators are evaluated by comparing the pressures inside the artificial ears with those developed by the same earphone inside 24 human ears. The problems associated with the design of an artificial ear are discussed. At the present state of knowledge concerning the physical properties of the human ear, the instrument of simplest design appears to be as usefull for calibration purposes as the more complicated devices.

Sommaire. On a déterminé les caractéristiques de quatre appareils d'étalonnage d'écouteur en comparant les pressions produites dans des oreilles artificielles avec celles produites dans 24 oreilles humaines pour un même écouteur. On analyse les problèmes que pose la réalisation d'une oreille artificielle. Dans l'état actuel de nos connaissances des caractéristiques physiques de l'oreille humaine, l'appareil d'étalonnage le plus simple de conception semble être aussi satisfaisant que des appareils plus compliqués.

Zusammenfassung. Die Wirkungsweise von vier Kopfhörer-Eichgeräten wird untersucht, indem man den Schalldruck innerhalb des künstlichen Ohres mit dem von demselben Kopfhörer in 24 menschlichen Ohren erzeugten vergleicht. Bei der Diskussion der mit der Konstruktion eines künstlichen Ohres zusammenhängenden Probleme ergibt sich, daß bei dem gegenwärtigen Stand unserer Kenntnisse über die physikalischen Eigenschaften des Ohres das Gerät mit dem einfachsten Aufbau ebensogut für Eichzwecke verwendbar zu sein scheint wie die komplizierteren Konstruktionen.

An artificial ear is an acoustic device designed to simulate the physical behaviour of the human ear by a network of massive, compliant, and resistive elements. It is used for calibrating earphones, the sound pressure developed by the earphone under test being measured with a microphone. If an "ideal" artificial ear and its associated electronic equipment were available, it would meet the following specifications:

a. Its input impedance would be the same as that of the average human ear so that the sound pressures inside the device would be identical with those developed in the auditory canal of the average real ear, whether that pressure be generated by an earphone or by a remote source.

b. Its response to transient and complex sounds over a wide range of pressure levels would be equivalent to that of the average human ear.

c. It would have sufficient stability to produce repeatable calibrations under uniform conditions.

Practical considerations such as cost and ease of manufacture, as well as acoustic instrumentation difficulties and the scarcity of information on the characteristics of the human ear, restrict modern artificial ears in their ability to meet these requirements. Another device widely used for calibration purposes is the earphone coupler, a rigid-walled cavity with a microphone mounted at one end. It is not designed to simulate the physical properties of the human ear.

In the past thirty years many different artificial ear and coupler designs have been proposed. A history of their development has been given by BARDUCCI [1]. Acousticians have expressed preferences among these designs which run along national lines. Five different devices for measuring the response of an earphone resting against the pinna have been accepted as tentative standards in Great Britain, France, Italy, Switzerland and the United States. Replicas of four of these instruments were constructed. Unfortunately, information on the French artificial ear was not available at the time this study was undertaken.

It was anticipated initialy that the designs could be compared by measuring their acoustic impedances, using the real ear as a reference. However, the input impedance of the *average* human ear as viewed from a telephone ear-cap is not readily determined. Large variations in this quantity have been reported in the literature [2], [3] and [4]. The input impedances of the four instruments were determined and all were found to lie within the reported range of values for the human ear.

Although the acoustic impedance is the more fundamental quantity, the sound pressure at some point in a system is more easily measured. A more definitive comparison of the several artificial ear designs was obtained by measuring the sound pressure levels under the ear-cap, using the pressure developed at the entrance to the auditory canal of the real ear for reference.

The measurements were made using two different earphones: a Western Electric model 711A which is a dynamic unit of superior frequency response, and a military type ANB-H-1 improved magnetophone. The constant power available and constant voltage

*) Now at U.S. Naval Postgraduate School, Monterey, California.

responses of these earphones were measured on the four calibration devices and on 24 human ears. The laboratory techniques employed were essentially those recommended by the American Standards Association [5]. The results of the applied voltage measurements are presented in Fig. 1 and 2.

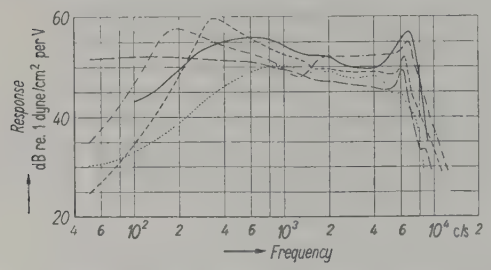


Fig. 1. Constant voltage response of Western Electric 711A earphone on four artificial ears and average human ear. Constant voltage to earphone 0.1 volt.
..... English artificial ear
– – – – Italian artificial ear
–·–·–·– Swiss artificial ear
— — — — American 6 cm³ earphone coupler
——— Average of 24 human ears.

Barducci [6] has posed six questions as being pertinent to the artificial ear design problem. The conclusions of this study will be presented as replies to these questions.

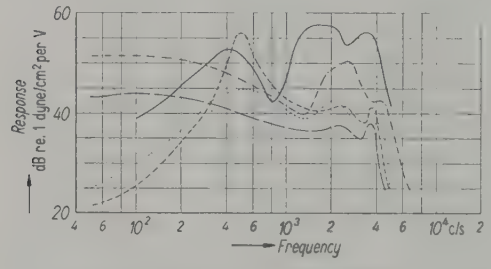


Fig. 2. Constant voltage response of Type ANB-H-1 earphon on four artificial ears and average human ear. Constant voltage to earphone 0.1 volt.
..... English artificial ear
– – – – Italian artificial ear
–·–·–·– Swiss artificial ear
— — — — American 6 cm³ earphone coupler
——— Average of 24 human ears.

1. *Should the seal between the receiver under test and the artificial ear be perfect?*

The variations in low frequency response due to leakage between ear-cap and pinna depend primarily upon the pressure exerted to hold the earphone to the head and the shape of the listener's outer ear. This leakage will be greater for hard bakelite earcaps than for soft rubber cushions because the greater elasticity of the latter provides a better seal. It appears that the seal should be perfect and that no attempt should be made to simulate this leakage until more quantitative data become available concerning the magnitude of these losses.

2. *What volume and shape should the artificial ear have?*

The volume should approximate that enclosed by the ear-cap on the real ear so that the sound pressure levels developed in the calibrator will be of the same order of magnitude as those generated in the human ear. This volume is about 6 cm³ for earphones which rest against the pinna. Unless an attempt is made to simulate the resonances of the human ear, the shape should be chosen so that the resonant effects are minimized. In the American 6 cm³ coupler, the influences of resonant modes are almost eliminated at frequencies below 6000 c/s.

3. *How should the earphone be attached to the artificial ear?*

So that the largest number of different types of earphones can be calibrated with the device. The attachment arrangement is most convenient in the British and Italian designs. The Swiss and American instruments do not have a projecting knife edge or a rubber gasket to simplify sealing the earphone to the device. Since the American design is so simple, an individual coupler can easily be constructed to fit each type of earphone to be calibrated.

4. *What value of acoustic resistance, if any, should be provided?*

Since the magnitudes of the damping elements associated with the auditory canal, pinna and tympanum of the average human ear are not accurately known, it is difficult to set up the specifications for the resistive elements which would be required to simulate these losses.

5. *Should a probe tube be used?*

Unless an attempt is made to duplicate the shape of the ear canal and to locate resistive elements in the appropriate positions, it appears that little is to be gained by using a probe tube.

6. *What frequency range should be explored?*

For telephone requirements: 300 ··· 4000 c/s. For special requirements: 50 ··· 10 000 c/s.

The design of an artificial ear involves quantities whose magnitudes are imperfectly known and phenomena that are incompletely understood. Until such time as more information on the human ear becomes available, the 6 cm³ rigid walled coupler would appear to be as useful for earphone calibration purposes as some of the more complicated devices.

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CONTRIBUTION AU PROBLÈME DE L'OREILLE ARTIFICIELLE

par A. Loncheval

Bell Telephone Manufacturing Company, Anvers

Sommaire. On utilise actuellement des coupleurs servant d'oreilles artificielles de types très divers et les résultats obtenus avec ces coupleurs présentent des écarts sensibles pour le même récepteur. Par conséquent les résultats obtenus dans divers laboratoires ne sont en général pas comparables. Dans le but d'obtenir des caractéristiques de récepteurs téléphoniques auxquelles on puisse se fier et qui soient comparables, on a mis au point une oreille artificielle ayant les mêmes caractéristiques que l'oreille humaine moyenne; tout d'abord l'impédance acoustique et la courbe de réponse aux diverses fréquences des oreilles humaines ont été déterminées. On montre pour terminer, que les résultats des mesures de caractéristiques de récepteurs téléphoniques, effectuées sur cette nouvelle oreille artificielle et sur l'oreille humaine moyenne sont pratiquement les mêmes.

Zusammenfassung. Durch die Benutzung vieler verschiedener Typen von künstlichen Ohren in den verschiedenen Laboratorien ergeben sich im allgemeinen merkliche Abweichungen in den Resultaten für gleiche Hörer. Um verläßliche Meßwerte für Kopfhörer zu erhalten, hat man ein künstliches Ohr konstruiert, das in seinen Eigenschaften den Durchschnittswerten des menschlichen Ohres angepaßt ist, wofür die Impedanz und der Druckverlauf im Ohr bei verschiedenen Frequenzen bestimmt wurden. Damit ergeben sich praktisch die gleichen Werte bei Messungen am künstlichen und am menschlichen Ohr.

Summary. With the variety of couplers used for artificial ears in practice, a wide difference in results is obtained with the same receiver which makes results of different laboratories not comparable. To obtain trustworthy response curves for telephone receivers an artificial ear having the same characteristics as the human ear has been constructed and the impedance and response curves of natural ears have been determined. These results agree with those of the new artificial ear.

1. Introduction

La méthode objective de mesure des caractéristiques aux diverses fréquences des récepteurs téléphoniques au moyen d'une oreille artificielle a suscité un intérêt croissant parce qu'elle permet de réaliser des progrès substantiels dans la communication téléphonique. Il existe actuellement de nombreuses oreilles artificielles qui ne possèdent pas toutes les mêmes caractéristiques et dont les caractéristiques sont différentes de celles de l'oreille humaine moyenne. À cause de ces différences:

a) il n'est pas possible actuellement de comparer les caractéristiques de réponse obtenues avec les différentes oreilles artificielles existantes, utilisées sur un même récepteur, puisque les caractéristiques de ces oreilles artificielles sont parfois très différentes;

b) les courbes de réponse obtenues avec un type d'oreille artificielle, utilisée sur des récepteurs de différents types, ne peuvent pas représenter la contribution exacte de cette partie du système téléphonique, si cette oreille artificielle ne possède pas toutes les caractéristiques essentielles de l'oreille humaine moyenne. Il s'avère donc nécessaire de posséder une oreille artificielle qui reproduit assez fidèlement toutes les caractéristiques physiques fondamentales de l'oreille humaine moyenne et du couplage avec les récepteurs. Aucune oreille artificielle réalisée jusqu'ici ne répond entièrement à ces exigences fondamentales; pour disposer de données suffisantes pour l'étude d'une oreille artificielle possédant les caractéristiques de l'oreille humaine moyenne, on doit connaître l'impédance acoustique d'entrée et comme complément la courbe de réponse de l'oreille humaine moyenne. Ces caractéristiques ont été déterminées au moyen de mesures faites sur les oreilles humaines. Tout d'abord, on a mesuré l'impédance acoustique des oreilles réelles, en employant la méthode de réaction d'une charge acoustique sur la source de son. Cette méthode a été décrite par R. D. Fay et W. M. Hall [1], [2] et est basée sur l'interréaction entre l'impédance électrique et la charge acoustique d'un transducteur électroacoustique; c'est en réalité une mesure de comparaison qui permet de déterminer une impédance acoustique inconnue en la comparant à une impédance acoustique connue et cela en mesurant

l'impédance électrique d'entrée du système total (transducteur + charge acoustique).

Des mesures ont été faites sur 20 oreilles humaines à des fréquences variant entre 500 et 3500 Hz; les valeurs de la résistance et de la réactance acoustique en fonction de la fréquence observées sur 20 oreilles avec un pavillon bien déterminé sont données par la courbe 2 de la Fig. 1. La caractéristique de réponse

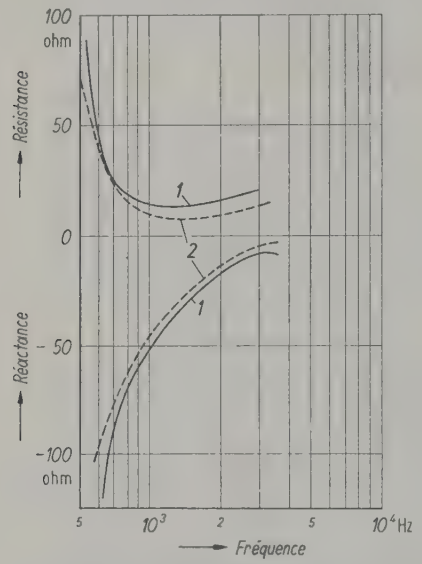


Fig. 1. Impédance totale de l'oreille humaine moyenne (2) et de l'oreille artificielle (1).

des oreilles humaines a ensuite été mesurée par l'intermédiaire de deux sondes acoustiques; l'une est introduite jusqu'au tympan et y mesure la pression acoustique; l'autre est fixée à la sortie des trous du pavillon d'un récepteur quelconque appliqué à l'oreille et permet d'y maintenir constante la pression acoustique. Les valeurs en décibel de la pression acoustique moyenne au tympan par rapport à la pression à la sortie des trous du pavillon, sont données par la courbe 2 de la Fig. 2.

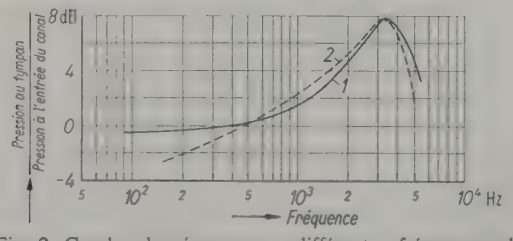


Fig. 2. Courbe de réponse aux différentes fréquences de l'oreille humaine moyenne (2) et de l'oreille artificielle (1).

2. Forme fondamentale de l'oreille artificielle

Celle-ci comportera deux cavités, le canal et le pavillon. Une seule cavité ne donnerait pas une impédance d'entrée correcte aux hautes fréquences; en effet, aux hautes fréquences, les dimensions du canal et du pavillon ne sont plus négligeables vis à vis des longueurs d'onde; on aura donc un déphasage entre la pression à l'entrée et à la sortie du pavillon et du canal; par conséquent, il n'est plus possible comme aux basses fréquences de remplacer l'action du coussin d'air dans le pavillon et le canal par un simple "stiffness". On devra dans ce cas tenir compte des propriétés des quadripoles en considérant le canal et le pavillon comme deux quadripoles en série avec leur impédance itérative [3].

On devra aussi tenir compte, dans la construction de l'oreille artificielle de l'absorption qui existe à l'intérieur de l'oreille, et de la fuite acoustique ou couplage imparfait entre l'oreille et le pavillon d'un récepteur téléphonique.

Comme informations permettant le calcul de la forme de l'oreille artificielle, on dispose, à part des résultats fournis par les courbes 2 des Fig. 1 et 2, des caractéristiques de l'oreille réelle moyenne et de la courbe de réponse du canal des oreilles réelles, mesurée par WIENER et ROSS [4].

3. Détermination du canal de l'oreille artificielle

La courbe de WIENER et ROSS donne la variation de la pression acoustique à l'intérieur du canal, en fonction de la fréquence (courbe 2 de la Fig. 3); la pression acoustique à l'entrée du canal est considérée comme constante. On peut considérer le tympan de l'oreille humaine comme une impédance mécanique complexe dont la masse peut être négligée devant la raideur. On peut très bien admettre que pour les fréquences envisagées dans cette étude, la raideur du tympan est suffisamment grande pour que son impédance soit aussi très grande par rapport à l'impédance itérative du canal et par conséquent le coefficient de réflexion du canal de l'oreille vaut environ l'unité.

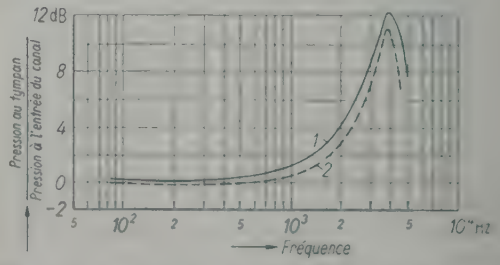


Fig. 3. Courbe de réponse aux différentes fréquences du canal de l'oreille humaine moyenne (2) (voir WIENER et ROSS [4]) et de l'oreille artificielle (1).

Si on assimile le canal de l'oreille à un tube terminé par une paroi rigide, celui-ci sera le siège d'ondes stationnaires. On peut donc choisir la pro-

fondeur du canal de façon à ce que son impédance s'annule à une fréquence bien déterminée. La courbe de WIENER et Ross donne une résonance vers 3800 Hz, ce qui correspond pour le tube choisi à une longueur de 23 mm. La réactance acoustique d'entrée de ce tube est évidemment fonction de sa section; le diamètre normal moyen du canal des oreilles réelles est de 7 mm. On peut en conclure que la réactance acoustique du canal des oreilles réelles correspond à la réactance d'un canal de 23 mm × 7 mm terminé par une paroi rigide. Un tube de 23 mm × 7 mm nécessite pour la mesure de la pression sonore, un microphone avec un diaphragme de 7 mm, dont l'impédance n'est pas infinie. Il a été jugé plus pratique d'utiliser un microphone commercial assez répandu comme le type 640 AA de la Western Electric. À cette fin on a calculé la réactance acoustique d'entrée d'un tube de 23 mm × × 7 mm ainsi que les modifications à apporter à ce tube pour y incorporer le microphone condensateur 640 AA.

4. Détermination des dimensions du canal

L'équation de l'impédance d'entrée d'un tuyau sonore ouvert à une extrémité et terminé par une impédance quelconque Z_0 est donnée par

$$Z_i = \frac{\varrho c}{S} \frac{\mathrm{j\,tg}\,(\omega l/c) + Z_0 S/\varrho c}{1 + \mathrm{j}\,(Z_0 S/\varrho c)\mathrm{tg}\,(\omega l/c)}. \qquad (1)$$

Si Z_0 est infini (paroi rigide), ce qui est à peu près le cas pour l'oreille humaine,

$$Z_i = -\mathrm{j}\,\frac{\varrho c}{S}\cot\frac{\omega l}{c}\,; \qquad (2)$$

$\omega = 2\pi f$ = pulsation,
ϱ = densité de l'air,
c = vitesse du son,
S = section du tuyau,
l = longueur du canal.

À partir de la formule (2), on a calculé la réactance d'entrée d'un tube de 23 mm × 7 mm; celle-ci correspond à la réactance d'entrée du canal de l'oreille humaine moyenne. Par l'adaptation d'un microphone condensateur à la sortie du canal de l'oreille artificielle, on abaisse considérablement sa réactance terminale ce qui correspondra pour le canal de l'oreille artificielle à un tube terminé par une réactance quelconque.

On déterminera alors les dimensions du nouveau canal à partir de la formule (1) si l'on connaît la réactance d'entrée du canal des oreilles et la réactance terminale que doit avoir le canal de l'oreille artificielle après adaptation du microphone condensateur: cette réactance d'entrée a été calculée et la réactance de sortie constituée par le diaphragme du microphone condensateur et l'espace d'air situé entre ce diaphragme et la sortie du tube a été mesurée (Fig. 4).

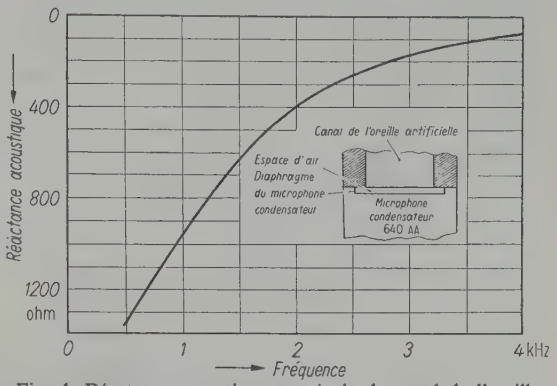


Fig. 4. Réactance acoustique terminale du canal de l'oreille artificielle.

En identifiant l'équation (1) à la réactance du canal des oreilles réelles, à deux fréquences suffisamment élevées, on en a finalement déduit un nouveau canal de 14 mm × 8 mm que l'on adapte pour le canal de l'oreille artificielle.

Remarque. Cette identification doit être faite à des fréquences assez élevées car à basse fréquence, le canal peut être remplacé par une simple capacitance acoustique et seul le volume de cette capacitance intervient pour donner une réactance d'entrée exacte; on pourrait alors trouver une infinité de dimensions pour le canal pourvu que le volume de celui-ci conserve sa valeur.

5. Réglage de l'absorption du canal

Une étoffe de velours à l'intérieur du canal peut donner une absorption suffisante. Le diamètre du canal étant conservé, on règle la surface du velours de façon à obtenir le même niveau que celui de la courbe de WIENER et ROSS (Fig. 3, courbe 2).

6. Détermination de la seconde cavité de l'oreille artificielle

À partir de la connaissance de l'impédance du canal de l'oreille artificielle que l'on peut maintenant mesurer, il est possible de déterminer les dimensions de la seconde cavité de l'oreille artificielle; cette cavité doit comprendre le volume d'air du pavillon de l'oreille réelle ainsi que l'espace d'air formé par la cavité extérieur du pavillon du microtéléphone. On peut considérer la seconde cavité de l'oreille comme un quadripole acoustique terminé par le quadripole du canal. On peut déterminer les dimensions de cette seconde cavité par la formule (1) que l'on identifie à l'impédance mesurée sur les oreilles humaines à deux fréquences de façon à isoler l et S. Ceci donne une cavité de 9 mm de profondeur

et de 22 mm de diamètre. Pour des raisons pratiques, (adaptation de l'oreille artificielle au pavillon du microtéléphone), on donnera à cette cavité une forme un peu différente, tout en lui conservant cependant la même hauteur (9 mm) et le même diamètre moyen (22 mm) (Fig. 5).

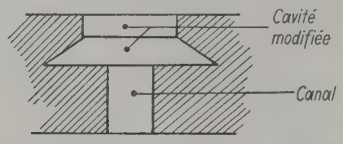


Fig. 5. Les deux cavités de l'oreille artificielle.

7. Réglage de l'absorption

La résistance acoustique de l'oreille artificielle doit coïncider avec celle des oreilles réelles. Celle-ci peut être réglée au moyen de la courbe totale de réponse (Fig. 2). Une résistance acoustique réglable en série avec la seconde cavité sera suffisante pour obtenir l'effet demandé.

8. Détermination de la fuite de l'oreille artificielle

Le couplage imparfait entre le microtéléphone et l'oreille humaine peut être reproduit par une fuite acoustique située au milieu de la seconde cavité. Cette fuite est constituée par un simple trou dont les dimensions sont à déterminer d'après la valeur de l'impédance de la fuite. En effet, le couplage imparfait provoque une fuite de pression acoustique qui donne avec la capacité du volume total de l'oreille artificielle une résonance à une fréquence bien déterminée et cela pour un même type de microtéléphone. De plus la résistance acoustique de cette fuite augmente très fortement la résistance acoustique totale à la fréquence de résonance. L'analogie électrique est donnée par la Fig. 6. La fuite doit donc être déterminée de façon à faire coïncider aux basses fréquences l'impédance de l'oreille artificielle et celle des oreilles réelles.

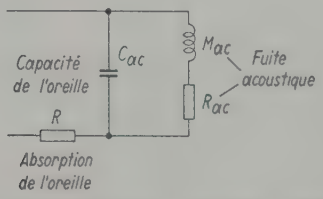


Fig. 6. Analogie électrique de l'oreille artificielle.

9. L'oreille artificielle

Elle comprend le microphone condensateur 640 AA et le coupleur composé des deux cavités: le canal dont les parois sont recouvertes de matière absorbante et le pavillon avec son absorption série et sa fuite acoustique. La Fig. 7 en donne une représentation.

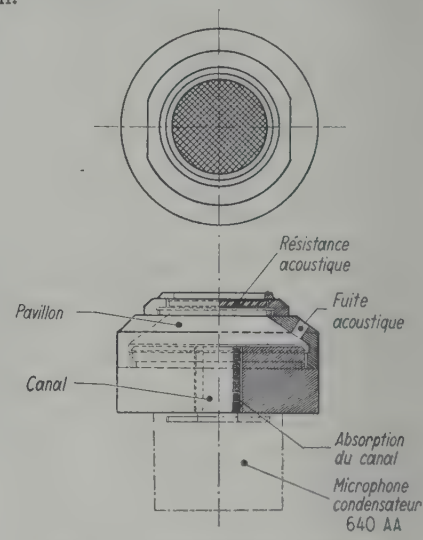


Fig. 7. L'oreille artificielle complète.

10. Caractéristiques de réponse

Les courbes de réponse aux différentes fréquences de l'oreille artificielle et de l'oreille humaine moyenne sont indiquées en Fig. 2; les courbes d'impédance sont montrées sur la Fig. 1. Ces courbes montrent clairement la concordance entre l'oreille artificielle et l'oreille humaine moyenne.

Les courbes de réponse de deux types de récepteurs ont été mesurées sur les oreilles moyennes et au moyen de l'oreille artificielle:

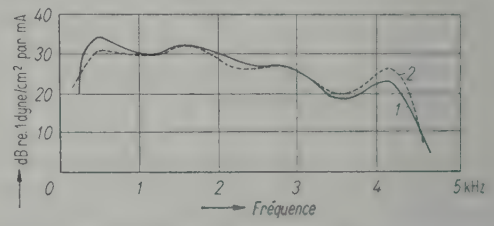


Fig. 8. Courbe de réponse du récepteur téléphonique équalisé dans un microtéléphone de type déterminé mesurée à l'entrée de l'oreille artificielle (1) et de l'oreille humaine (2).

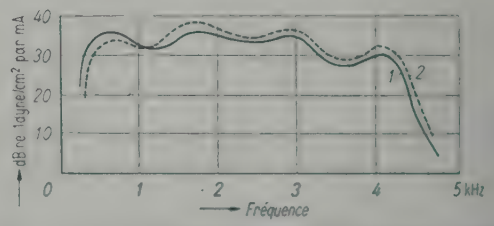


Fig. 9. Courbe de réponse du récepteur téléphonique équalisé dans un microtéléphone de type déterminé mesurée avec l'oreille artificielle (1) et avec l'oreille humaine (2).

a) à l'entrée de ces oreilles (Fig. 8) pour un récepteur type équalisé;

b) au tympan (Fig. 9 et 10) pour un récepteur type équalisé et non équalisé.

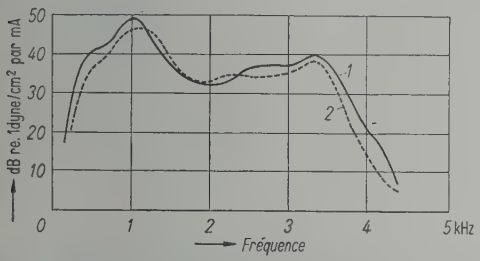


Fig. 10. Courbe de réponse du récepteur télephonique résonant dans une microtéléphone de type déterminé mesuré avec l'oreille artificielle (1) et avec l'oreille humaine (2).

On peut de nouveau observer ici une correspondance très convenable.

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Discussion

Morton, J. Y.: I think the impedance of the ear looking into the entrance of the ear canal cannot be represented by a tube terminated in a rigid wall at the position of the drum, with absorption at the walls of the tube, because the majority of the absorption comes from beyond the ear drum. (A slide was shown giving the mean impedance of 6 normal ears as seen from the tip of an insert ear-mould to justify this point.) The impedance from the ear drum inwards can be represented by three series acoustical elements, resistance, mass and compliance (see discussion on Mr. W. W. Lang's paper in Section IV for impedance values between 200 and 4000 c/s). If the tube was terminated in elements simulating the ear drum impedance a probe tube microphone placed in front of these elements could be expected to give the sound pressure in front of a human ear drum; the condenser microphone, because of its impedance not being that of the ear drum, can not be expected to do this. Should not the resistance in the equivalent pinna cavity of the artificial ear be a bridging element such as an acoustical resistance tube in the side of the cavity and not a series element as it appeared to be, for it is there to represent the absorption at the walls of the pinna.

Conférencier: Il est exact que l'absorption du canal est située en grande partie au tympan de l'oreille réelle donc en série avec cette cavité, mais dans la réalisation de l'oreille artificielle, il est plus pratique de placer cette absorption sur les parois du canal, donc en parallèle avec cette cavité. Ceci peut se faire sans nuire au résultat final pour autant que l'absorption soit suffisante pour donner à la courbe de réponse du canal un niveau équivalent à celui donné par la courbe de Wiener et Ross, et pour rendre exacte l'impédance d'entrée du canal. Il en est de même pour la cavité du pavillon de l'oreille artificielle pour laquelle une résistance placée en série s'avère plus pratique.

SPEZIELLE FRAGEN BEI DER SPRACHGEHÖRPRÜFUNG UND HÖRHILFENANPASSUNG

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Zusammenfassung. Beim Aufbau von Wortlisten für Gehörprüfungszwecke benutzt man hauptsächlich zwei Prinzipien, und zwar a) den Formalismus der gleichen statistischen Lauthäufigkeiten für Test- und Normalsprache und b) das Prinzip der lautanalytisch, d.h. spektroskopisch gleichen Wortgruppen. Diese Auswahlprinzipien werden diskutiert und einige neue Resultate für die deutsche Sprache angegeben.

Summary. Word-lists for hearing tests can be constructed in two principal ways: firstly, the formalism of equal statistic distribution of phonetic sounds in the words of the test and in normal speech, and secondly the principle of finding equal word-series by frequency-analysis and sorting the words. These principles are discussed and some new results are given for the German language.

Sommaire. La constitution des listes de mots servant aux essais phonométriques se fait d'après les deux principes suivants: a) on cherche à obtenir dans l'essai pour chaque son élémentaire la même fréquence statistique que dans la parole courante; b) on cherche une même composition spectrale des groupes de mots. On étudie ces deux principes et on donne quelques résultats nouveaux relatifs à la langue allemande.

1. Allgemeine Gesichtspunkte für den Aufbau von Prüftesten (Wortauswahl)

Eine exakte Sprachgehörprüfung und Hörhilfenanpassung bedarf sorgfältig aufgebauter Prüfteste. Bei der Aufstellung von Testwortlisten geht man von zwei Grundsätzen aus: Die benutzte Sprache soll:

a) in ihrer Zusammensetzung dem Durchschnittswortschatz des Prüflings angepaßt sein, d.h. die gleiche Lauthäufigkeit wie seine Sprache besitzen;

b) in der Verteilung der Sprachlaute eine passende

Mannigfaltigkeit zeigen. Mit einer Erweiterung dieses letzteren Gedankens auf Grund elektroakustischer Wortanalysen haben wir den Kern des eigentlichen Problems bei der Sprachgehörprüfung getroffen.

Es läßt sich nicht ohne weiteres sagen, ob beide Auswahlgrundsätze zu kongruenten Ergebnissen führen. Sie sind grundsätzlich heterogener Natur und es bedarf sorgfältiger Wortauswahl, um beide Prinzipien einigermaßen zu befriedigen.

Die Wortauswahl nach Lauthäufigkeiten ist zuerst von amerikanischen und französischen Autoren [1] benutzt worden. Man unterwirft bei diesem Auswahlschema die große Zahl der vorkommenden Sprachlaute einer Häufigkeitsbewertung. Nach dieser Bewertung wird dann das Testmaterial sortiert, d.h. die Häufigkeit, mit der ein Laut in der normalen Umgangssprache vorkommt, soll auch im Prüftest erhalten bleiben. Dieser Auswahlgesichtspunkt ist also rein lautstatistischer Natur.

Eine speziell für Wortlisten der Sprachgehörprüfung durchgeführte Lautstatistik gab es in Deutschland nicht. Wohl hatte HELMUTH MEIER [2] eine umfassende Lautzählung und Häufigkeitsuntersuchung (100 000 Laute) der bühnenhochdeutschen Sprache durchgeführt. Sie könnte, wie das HAHLBROCK versucht hat, für die Sortierung zugrunde gelegt werden.

Abb. 1 zeigt eine von uns für die Kurzworte gefundene geordnete Häufigkeit der Sprachlaute (dick ausgezogene Linie). Daneben wurden für die gleiche angenommene Lautbasis die Werte der Häufigkeiten der MEIERschen Zählung für 100 000 Laute eingetragen. Bis auf die Laute *d* und *n* stimmen beide Lautreihen mit einer mittleren Streubreite von 0,9% überein. Im Satz und im längeren Wort kommen die Laute *d* und *n* offenbar wesentlich häufiger vor als im Kurzwort.

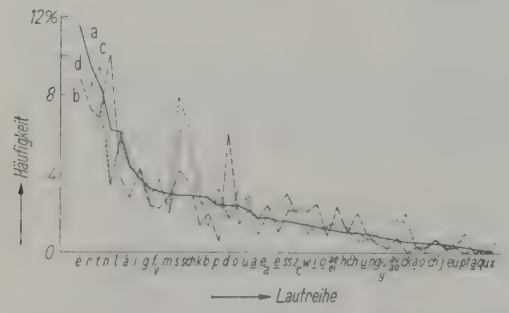


Abb. 1. Relative Häufigkeiten der Sprachlaute in deutscher Umgangssprache und mit Prüftest.
a) ein-zweisilbige Worte westdeutscher Aussprache (30 000 Laute, Aufnahmen des Akustischen Laboratoriums der Med. Akademie Düsseldorf).
b) Vergleichswerte der Häufigkeitszählung nach H. MEIER (100 000 Laute aus der hochdeutschen Bühnensprache).
c) Häufigkeitsverteilung in einem auf isophone Komponenten aufgebauten Gehörprüftest.
d) Häufigkeitsverteilung im verkürzten Test (90 einsilbige Worte).

Wir wollen nun eine solche Häufigkeitsreihe nur für die Vorwahl der gesamten in dem Prüftext auftretenden Laute benutzen. Der eigentliche Aufbau der Wortliste erfolgt mit Hilfe elektro-akustischer Analysiermittel. Hierdurch erst kann eine passende Mannigfaltigkeit der Wortfolge erzielt werden. Die Lauthäufigkeitsfolge erleidet jedoch dabei gewisse Verschiebungen. Die beiden Kurven c und d der Abb. 1 zeigen diese Veränderungen im Prüftext gegenüber der Normalsprache an.

Schon bei den ersten Benutzern von Wortlisten für Gehörprüfung waren andere Gedanken hervorgetreten und besonders in der LAMPERTschen Wechselreihe [3] recht eindrucksvoll durchgeführt worden. Einer dieser Gedanken war der, alle Teiltongebiete der Sprache „möglichst gleichmäßig" zu erfassen.

Beim Aufbau von Gehörprüftesten ist zweifellos die Erzielung einer „gleichmäßigen" Belastung der einzelnen Bereiche der Basilarmembran ein wesentlicherer Gesichtspunkt als die Erfüllung des Häufigkeitsprinzips.

Daneben sind psychologische Auswahlgesichtspunkte oft nicht unerheblich. Das Wort soll ja für jeden Prüfling dem Inhalte nach gleich gut verständlich sein. Die Benutzung gleich schlecht verständlicher Worte, z.B. sinnloser Silben, hat sich in Deutschland nur in der Fernsprechprüftechnik bewährt. Für die audiologische Prüfung werden Logatome in deutschen Hörprüfinstituten jedoch kaum benutzt.

2. Die elektroakustische Auswahlmethode

Die Forderung an die Wortliste nach einer bestimmten maximalen Wechselbelastung des Ohres in den einzelnen Frequenzbereichen bedeutet eine Wortauswahl nach frequenzanalytischen Gesichtspunkten [4], also eine Auswahl mit Hilfe elektroakustischer Siebmittel. Betrachtet man im Visible Speech-Bild [5] nur die Verteilung der Intensitätsmaxima in Abhängigkeit von der Frequenz, aber unbekümmert um ihre zeitliche Folge, so lassen sich bei einsilbigen Kurzworten, und nur diese sollen hier im Augenblick als Testworte betrachtet werden, vier Worttypen unterscheiden.

In der Abb. 2 sind je zehn Worte jeder Type zusammengestellt. Jede Type zeigt also eine spezifische Frequenzkurve der Hauptintensitätskomponenten, trotzdem die Worte der einzelnen Gruppe durchaus nicht immer sprachlich oder klanglich als gleichartig gelten können. Je nach Wahl einer solchen Gruppe wird das Ohr längs der Frequenzskala verschieden

stark belastet. Die Schalldruckunterschiede in den einzelnen Spektralbereichen betragen im Mittel etwa 7 dB, beim einzelnen Wort der Gruppe jedoch bis zu 20 dB.

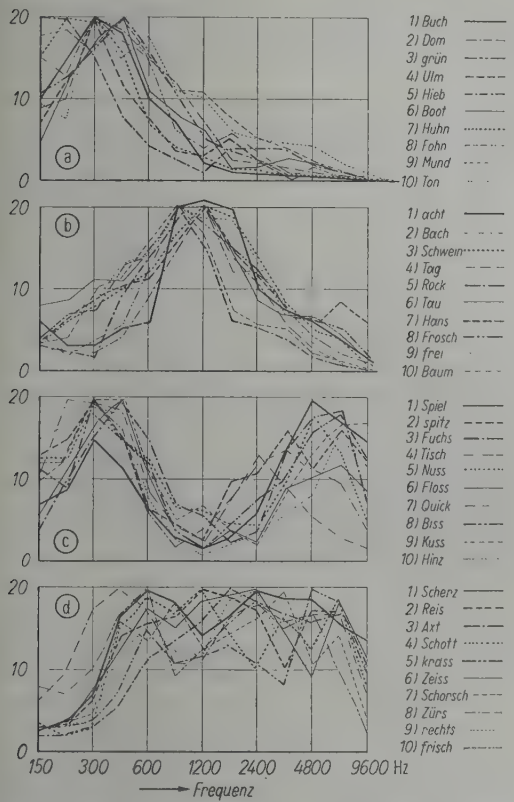


Abb. 2. Die vier hauptsächlich vorkommenden spektralen Verteilungen im gesprochenen Wort.

Wenn nur der Hörschaden, nicht aber die Wortauswahl das Ergebnis der Hörprüfung beeinflussen soll, muß jede Wortgruppe des Prüftestes so aufgebaut sein, daß alle Teiltongebiete gleichmäßig angesprochen werden, d.h. in jeder gleichlaut wiederzugebenden Wortgruppe sind die vier Grundtypen der Abb. 2 entsprechend zu verteilen. Abb. 3 mag ein solches Verteilungsschema für einen Hörprüftest darstellen. Für jede der vertikalen gleichlauten Wortgruppen wechselt zwar im einzelnen Wort die Ohrbelastung beträchtlich, doch sind alle Spitzenwerte der ganzen Wortgruppe so geordnet, daß sie eine um etwa $^1/_2$-Oktave versetzte Folge gleichgroßer Belastungen bilden. Bei größerer Wortzahl je Gruppe kann die Folge auch noch dichter (z.B. $\frac{1}{4}$-Oktavabstand) gemacht werden. Diese Folge der Maxima ist am Kopf jeder Gruppe angegeben. Die vertikalen Kolonnen sind also hinsichtlich der Ohrbelastung gleichwertig und können untereinander ausgetauscht werden. Das ist besonders vorteilhaft für die Sprachgehörprüfungen mit verschiedenen Lautstärken (Verständlichkeitsdiagramm). Wir haben zur Gewinnung von Prüftexten für die Sprachaudiometrie bisher etwa 400 einsilbige, einfache deutsche Worte bei einer definierten Sprachlautstärke von 50 Phon auf diese Weise spektroskopisch geordnet und eine ganze Reihe solcher Sprachgehörprüfblätter im Laufe des letzten Jahres entwickelt.

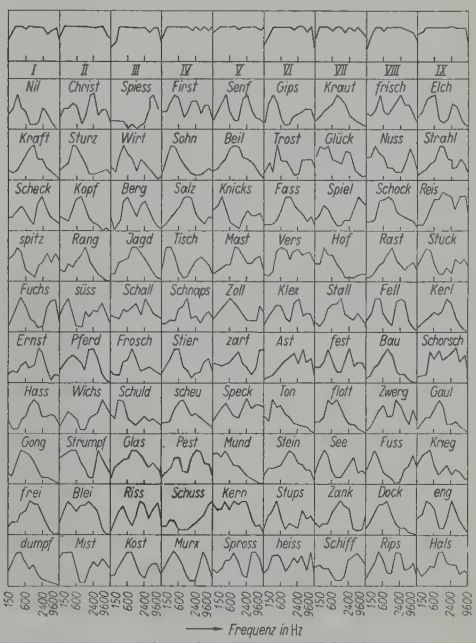


Abb. 3. Gekürzte Liste eines Verständlichkeitstestes für Gehörprüfung mit den zugehörigen Frequenzkurven der Hauptwortkomponenten (für jede vertikale Kolonne ist die Belastung der einzelnen Spektralbereiche konstant).

Hängen solche Wortbilder der spektralen Hauptkomponenten nun stark von der Art des Sprechens ab? Die Frequenzkurven der Intensitätsmaxima ändern sich im Gegensatz zum Visible Speech-Spektrogramm nicht stark, wenn der Sprecher die Aussprache unwesentlich verändert oder wenn man abwechselnd einen männlichen oder weiblichen Sprecher mit nicht zu stark differierendem Grundton im Stimmklang nimmt. Abb. 4 mag die Änderung an den beiden Worten *Ast* und *Maus* illustrieren. Die Worte wurden mit vier verschiedenen natürlichen Lautstärken gesprochen. Es zeigte sich, daß die schwach vokalisch gefärbte Aussprache mit etwa 40 ··· 45 Phon Lautstärke die größten Unterschiede ergibt, und daß die normale Lautstärke von 50 Phon, bei der die Vokale schon deutlicher hervortreten, nur eine geringe spektrale Differenzierung der Maxima erzeugt. Selbstverständlich beziehen sich diese Angaben auf Sprecher mit einwandfreier Artikulation und klarer nicht zu hoher und nicht zu tiefer Stimme. Als Folgerung dieses Vergleiches darf man die Laut-

stärke von 50 Phon als besonders geeignet für die Bandbesprechung für Hörprüfzwecke bezeichnen.

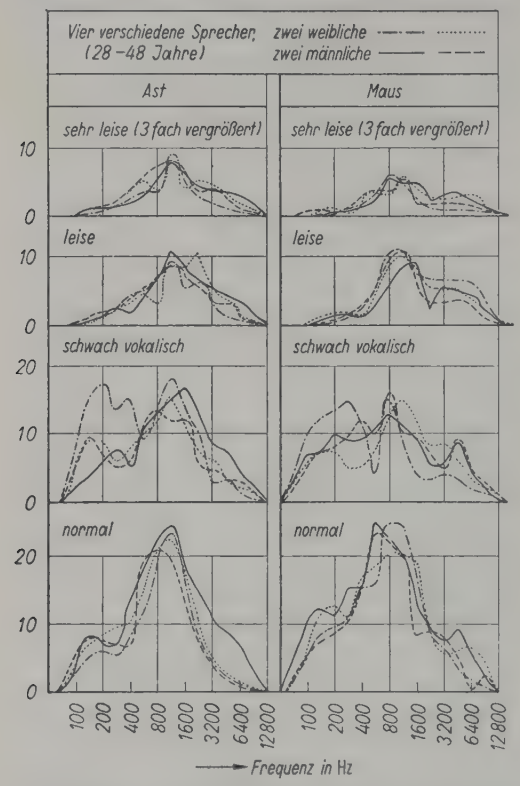


Abb. 4. Einfluss verschiedener Sprecher und verschiedener Lautstärke auf das Spektrum der Hauptkomponenten zweier Worte.

Es wäre noch nachzutragen, in welcher Weise sich die Häufigkeitskurve der Sprachlaute bei konsequenter Benutzung des spektroskopischen Ordnungsprinzips verändert. Es zeigt sich, wie man aus Abb. 1 erkennt, daß die stimmlosen Konsonanten und Zischlaute, insbesondere die Laute *s* und *sch*, die zur Anhebung der hohen Frequenzen häufiger im Prüftest benutzt werden, nun aus der Rangreihe der Laute etwas herausfallen.

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Diskussion

Mrass, H.: Vom neurophysiologischen Standpunkt wäre es zweckmäßiger, bei dem Siebverfahren zur Gewinnung einer gleichmäßigen Belastung der Basilarmembran Melfilter zu verwenden.

Vortragender: Die Verwendung einer Mel-Breite des Filters gegenüber der Oktavbreite dürfte für die Sprachanalyse, ähnlich wie bei den Geräuschanalysen, Bedeutung gewinnen. Halboktavfilter sind u.U. noch günstiger. Die Sprache zeigt eine gewisse Variationsbreite, wodurch die zweckmäßigste Bandbreite nicht so einfach feststellbar ist. Ich halte die Benutzung der Mel-Breite nicht für erforderlich.

THE DESIRED PERFORMANCE OF HEARING AIDS

by H. Mol
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Summary. The author draws attention to the fact that the success of the application of a hearing aid depends on the dynamic properties of the ear. The ossicle chain must be regarded as a regulating mechanism that is brain-controlled by means of the stapedial muscle. Special difficulties arise when the dynamic range of the basilar membrane itself is diminished. In principle a hearing aid with automatic gain control can never fully restore the impaired dynamic behaviour of the ear.

Sommaire. L'auteur attire l'attention sur le fait que l'efficacité d'un appareil de prothèse auditive dépend des caractéristiques dynamiques de l'oreille. Il faut considérer la chaîne des osselets comme un mécanisme régulateur qui est commandé par le cerveau au moyen du muscle stapédien. Il se présente des difficultés spéciales quand les qualités dynamiques de la membrane basilaire elle-même sont diminuées. En principe, un appareil de prothèse auditive, avec réglage automatique du gain, ne peut jamais corriger complètement un comportement dynamique défectueux de l'oreille.

Zusammenfassung. Der Autor macht darauf aufmerksam, daß der Erfolg bei der Anwendung von Hörhilfen von den dynamischen Eigenschaften des Ohres abhängt. Die Reihe der Gehörknöchelchen muß als Reguliermechanismus betrachtet werden, der mit Hilfe des Steigbügelmuskels vom

Gehirn beeinflußt werden kann. Besondere Schwierigkeiten entstehen, wenn der Dynamikbereich der Basilarmembran selbst vermindert ist. Prinzipiell kann ein Hörgerät mit automatischer Verstärkungsregelung niemals vollkommen die beeinträchtigten dynamischen Funktionen des Ohres wiederherstellen.

It is impossible to discuss the desired performance of a hearing aid without referring to some properties of the ear.

There are several reasons for which a hearing aid can never fully compensate deafness. The biggest difficulties arise when the basilar membrane is abnormal. In that case the location of the hearing aid is fundamentally wrong: it provides amplification in the sound conducting system whereas the deficiency is in that part of the ear which transforms the mechanical vibrations into nerve impulses.

It is often overlooked that the ear contains an automatic gain control in the ossicle chain. Strictly speaking one ossicle between the tympanic membrane and the oval window would be sufficient for matching purposes, as in birds.

The dynamic range of the basilar membrane, however, is surprisingly small. The basilar membrane is already overloaded by excursions of the stapes some 30 dB above threshold. We have been able to establish this fact in numerous ears that were bereft of the middle ear mechanism, as is the case after the fenestration operation or radical mastoidectomy.

The automatic gain control has a feedback character: the stapedial muscle can regulate the ratio of the ossicle chain by changing the axis of rotation of the stapes. The ratio of the ossicle chain is defined here as the relation between the volume displacement of the stapes and the excursion of the tympanic membrane.

As shown in Fig. 1 there are two main axes of rotation of the stapes: one is parallel to the short axis of the footplate but goes through one of its poles. Rotation around this axis (a) is facilitated by the anatomic configuration. Because of the pedal-like movements of the footplate rotation around axis (a) is very efficient in displacing the fluid in the cochlea (Fig. 2a).

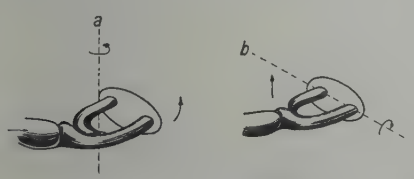


Fig. 1. The two main axes of rotation for the stapes.

A second extreme mode of vibration is rotation around the long axis of the footplate (b). Because of symmetry there is a compensating current near the stapes: the displacement of the inner ear fluid is rather ineffective.

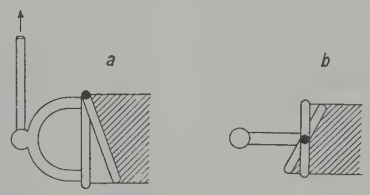


Fig. 2. The axes of rotation of the stapes and the displacement of the cochlear fluid.

By contraction the stapedial muscle favours rotation around axis (b) and resists rotation around (a). In other words: the ear becomes less sensitive. Now the feedback feature of the system is that the contraction of the stapedial muscle is derived from the nerve output because that muscle is brain controlled.

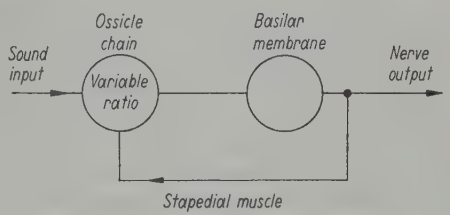


Fig. 3. The ear as a feedback system.

This regulating mechanism is rather slow and reacts to average values rather than to peak values. That is why in normal speech the basilar membrane must cope with the peaks that slip through the automatic gain control of the middle ear.

Even people without a middle ear system have no difficulties with the peaks of amplified normal speech. In general people with deafness of the conductive type have no problems with hearing aids. As long as the basilar membrane is healthy there are no dynamic difficulties though the regulating mechanism may be far less efficient than in the normal ear.

In many cases of perceptive deafness, however, the dynamic behaviour of the ear is severely impaired because the dynamic range of the basilar membrane has become very small. Patients with perceptive deafness are likely to have dynamic difficulties with a hearing aid.

Apparently there are two main causes for a diminished dynamic range of the ear in deafness:

a. the mechanical condition of the variable element, the ossicle chain is impaired by stiffness.

This results in a hearing loss and also in a diminished dynamic range;

b. the basilar membrane has undergone pathological changes of a certain type, also resulting in a hearing loss and a diminished dynamic range.

In the latter case the patient has troubles with a hearing aid whereas in the former case he has not.

Strictly speaking a hearing aid with an automatic gain control can never fully compensate the decreased dynamic range of the basilar membrane because the hearing aid is in front of the impaired regulating system whereas it should be in series with it. Automatic gain control in a hearing aid can, however, give the patient a more comfortable hearing. Still it remains a difficult problem to satisfy people with perceptive deafness by a hearing aid.

In general the frequency curve of a hearing aid should have a smooth character. There should be no prominent peaks in the response. Even in modern telephone apparatus where one can reckon with normal ears the frequency curves are kept smooth in order to avoid a harsh reproduction. Especially the ears with perceptive deafness are sensitive to peaks in the frequency response. In particular peaks in the formant regions should be avoided.

In my opinion, before fitting a hearing aid one must know if the basilar membrane is healthy or not.

There are several methods for investigating the condition of the basilar membrane. In our opinion even the threshold audiogram for air and bone conduction as determined by a reliable audiometer using a standardized technique is a suitable means. The fault is not with the audiogram but with the person who interprets the audiogram. In doubtful cases it is advisable to determine, in addition, the dynamic range of the ear in question.

The idea of fitting a hearing aid in the same way as fitting a pair of spectacles has at last been abandoned, though not by all. Attempts to give selective amplification compensating the loss shown by the threshold audiogram have failed. There are several reasons for that. Primarily the behaviour of the ear at threshold levels may substantially differ from that at higher intensities. Secondly sharp frequency cut-offs of perceptive origin cannot be compensated by a complementary amplification in the mechanical system because of the transients that are introduced in that way. Thirdly, if certain parts of the basilar membrane have undergone pathological changes it is wise to leave them in peace and not to amplify these regions. Sometimes it proves even to be better to cut off these frequencies in the hearing aid.

It is a matter of experience that a patient always needs less amplification than the threshold audiogram suggests. One can expect this because in general his dynamic range is smaller than in normal hearing. In this respect the fenestrated ear is interesting. After the fenestration there usually remains a hearing loss of some 30 dB at threshold. Sounds of normal intensity, however, appear louder to the fenestrated ear than to the normal ear.

If a patient has been deaf for a long time he cannot remember what was natural. He will prefer the hearing aid that amplifies only the frequencies he used to hear in his deafness. In general this will not be the aid that gives him the maximum articulation.

In order to be able to control the smoothness of the frequency response of the hearing aid it is necessary tom standardize the method of measurement. Of course one should choose the method that is best adapted to the normal way of using the hearing aid. In the authors' opinion this is the free-field method.



SECTION V
ULTRASONIC ELECTRO-ACOUSTICS

GENERAL REVIEW

by G. Bradfield
National Physical Laboratory, Teddington, England

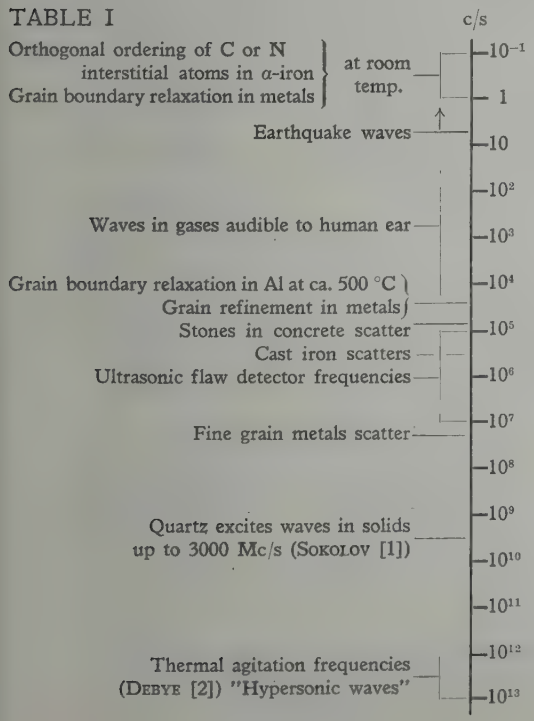


1. Introduction

In reviewing electro-acoustics in the ultrasonic frequency range it will be found necessary also to consider certain aspects of launching and propagation, for these subjects are closely linked with it.

Examination of features of the frequency range of mechanical waves in Table I reveals at the low frequency end very slow processes such as the orthogonal ordering of solute atoms and, at the other end, frequencies of the order of 10^{13} c/s where mechanical waves begin to merge indistinguishably into electro-magnetic waves. At these frequencies, though the term ultrasonic is well-justified, the waves concerned are more often termed hypersonic or thermal agitation waves. While they can be employed, through the medium of X-ray wave diffusion, to measure the elastic constants of solids, this use is not strictly electro-acoustical and the highest frequency to which this term so far seems applicable is 3000 Mc/s up to which, according to Sokolov [1], quartz can function satisfactorily as an electro-mechanical transducer.

TABLE I



It will be noted that the frequency range depicted from sub-audible to hypersonic embraces 12 decades, of which the audible range constitutes rather over 3 decades, and the ultrasonic 5 decades or more. From the electro-acoustic point of view the distinction between these two parts is artificial. However, it will be obvious that the range of even 5 decades makes it difficult to cover the subject in a brief review, for what is true for one part may not apply to another. This short review therefore will inevitably omit much which may well be termed noteworthy, for it will be necessary to survey the field very broadly. Publications by Mason [3], [4] (1948 and 1950) and by Bergmann [5] (1949) form valuable sources of information on the subject up to about 1949.

2. Classification of transducers

It is of assistance to classify electro-mechanical transducers as in Table II. The separation into reversible and irreversible categories is accompanied by the distinction that the principle of reciprocity [6], [7] is applicable to the former, but not to the latter. Of these, the spark [8] is noteworthy because its frequency spectrum is fairly uniform and extends into the region of megacycles per second. The methods in this category are merely electrical methods of measuring heat and are marginal for inclusion as electro-acoustic transducers. The last item, frequency modulation, refers to the use of mechanical vibratory motion to alter the capacity of an electrical condenser governing the frequency [9] of an oscillator, a method employable with advantage up to frequencies of megacycles per second.

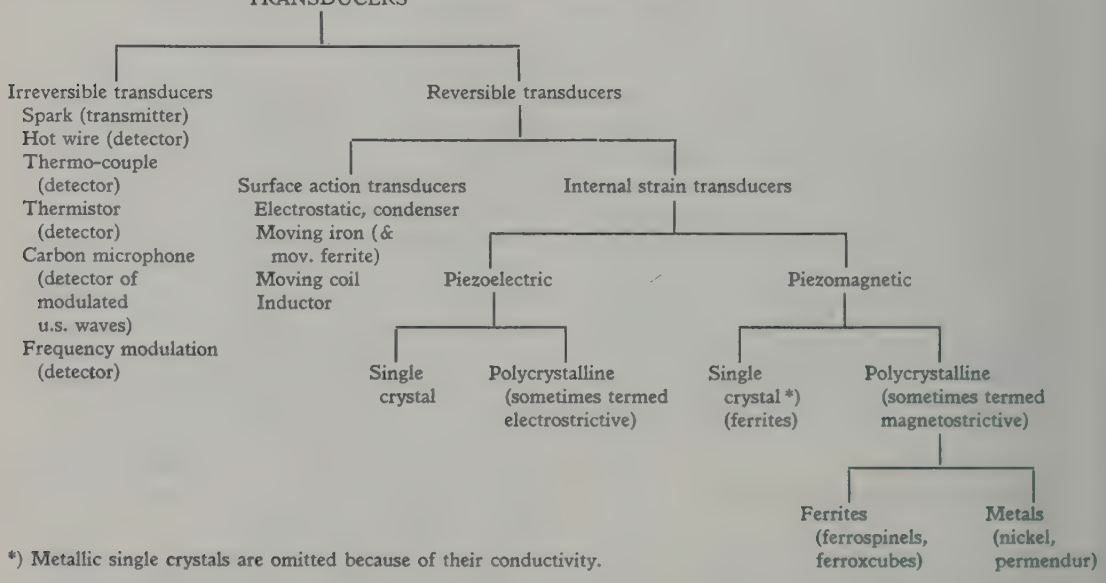
The transducers which are here termed surface action transducers, have been used for many years, but new techniques have enabled the electrostatic, the inductor, and perhaps the moving "iron" in the form of ferrites to function even into the 100 kc s region and above.

In the study of ultrasonics, however, megacycle frequencies are important, and here surface action transducers deteriorate in performance and leave the field to transducers of the internal strain type, more particularly known as "piezoelectric" and "piezomagnetic" (this term seems preferable to "magnetostrictive") where every atomic group of the material can be stressed almost with the speed of light by the application of either electric or magnetic fields.

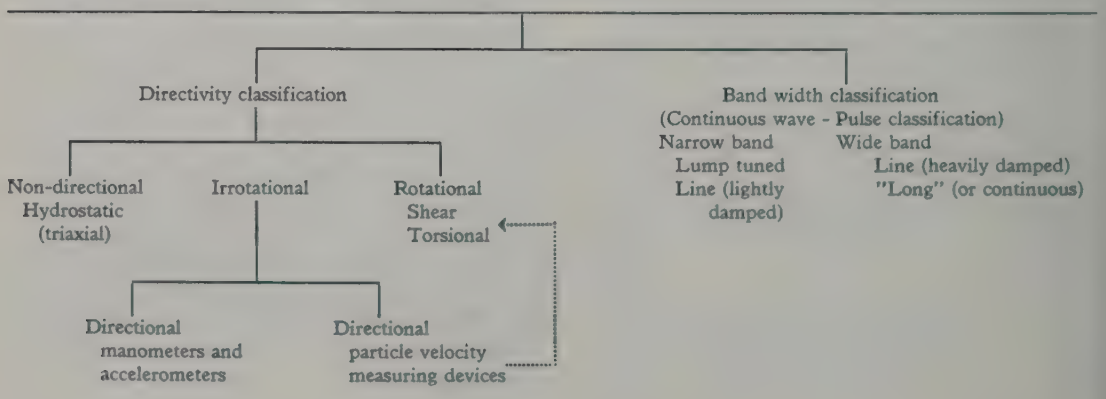
Looking back some twenty years, quartz undisputedly held chief place as a transducer of this type,

TABLE II

CLASSIFICATION OF ULTRASONIC ELECTRO-ACOUSTIC TRANSDUCERS



*) Metallic single crystals are omitted because of their conductivity.



and there still exist certain special applications such as the quartz clock [10] and Loran [11], for oscillators of extremely good frequency stability — of the order of 1 in 10^9 — where quartz is unrivalled, and for normal frequency stabilisation purposes — of the order of 1 in 10^6 — it still holds the field. Yet it is encountering formidable rivals, for example, in ethylene diamine tartrate for electro-mechanical filters (see also [26]) and in barium titanate [26] for the generation of medium and high intensities of ultrasonic waves. Furthermore, materials such as rochelle salt, ammonium dihydrogen tartrate and again barium titanate can be shaped to perform functions for which quartz is unsuitable. There exists also the alternative of magnetostrictive materials which have been widely used for echo-sounding and low frequency work, and here nickel is giving place to *permendur* (49% Co, 49% Fe, 2% Va) though the position of both is threatened by barium titanate. Ferrites must also be discussed, for these show great promise, for instance, as electro-mechanical filter elements [12].

Table II in its lower part contains a division of transducers in terms of their directivity, a classification which becomes more complex in the ultrasonic field, for, in contrast to the usual situation in the audible range, the tangential motion of surfaces is often of consequence. Directional manometers and accelerometers are available, while, for particle velocity measurement, thin threads can be applied to surfaces and can permit resolution of the normal and tangential components. By the use of several such devices shear and torsional distortion can be investigated, a connection shown dotted in Table II.

3. New polycrystalline transducer materials

The increasing importance of polycrystalline transducer materials is noteworthy. These materials are, in the piezoelectric field, barium titanate and its derivatives, and in the piezomagnetic field, ferrites e.g. double oxides of the $a\beta O \cdot b\gamma O\ Fe_2O_3$ type often called *ferroxcubes* and *ferrospinels*, β and γ being certain divalent metals. A typical composition is (90% NiO, 10% ZnO) Fe_2O_3 [13]. As will be evident from this brief survey, they may be said to be producing a revolution in the field of electro-acoustics, at least so far as ultrasonic frequencies are concerned, and it is worth while to consider why this is so and what advantage can be taken of the new situation. Both of these are ceramic materials made by similar high temperature (circa 1250 °C) firing processes; they are hard, non-conducting, brittle, substantially nonhygroscopic and can be provided with conducting surfaces, strongly bonded where necessary by additional high temperature firing processes. The raw materials used are cheap and plentiful, and the mode of manufacture can provide even complex shapes quite cheaply where large-scale manufacture is involved. The second feature to be noted is that these materials have to be polarised before they have any appreciable electro-mechanical coupling. This polarisation process may or may not be durable. In the case of certain barium-titanate modifications containing lead titanate the coercive force [14] is very high, and certainly for such compositions there is no need to provide continuous polarisation by external means. They can even be cut and ground without destroying the polarisation. Even without the lead, barium titanates with normal usage show little deterioration in performance after the first few days following polarisation. In the case of the ferrites it is generally, though not invariably, advantageous to provide means to maintain this polarisation. The polarisation process aligns adjacent regions or domains normally random in their directions of polarisation and VON HIPPEL [15] has described in a very elegant way the mechanisms of the domain wall movements. The incremental permittivity and permeability respectively are governed by the ability of local regions to change polarisation quickly and reversible, and for both classes of material the behaviour up to about 10^7 c/s with frequency of alternation of the field is rather similar, as has been shown in Fig. 1 taken from the work of VON HIPPEL [16].

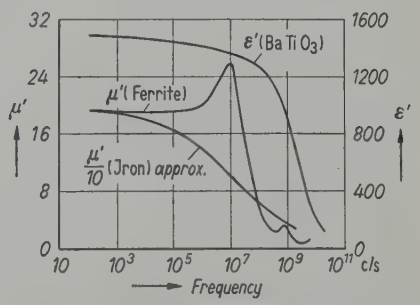


Fig. 1. The incremental permittivity and permeability as a function of frequency.

The properties resulting from these polarisation processes are so similar in the two classes of materials, e.g. titanates and ferrites, that the electro-mechanical coefficients conform to identical matrices as shown below:

0	0	0	0	g_{15}	0
0	0	0	g_{15}	0	0
g_{31}	g_{31}	g_{33}	0	0	0

These, it will be observed, correspond to the crystalline class of dihexagonal pyramidal symmetry. It is also to be observed that these constants vary quite widely with composition, the grain size

of constituents, firing temperature and atmosphere, rate of cooling and, of course, with the polarisation process used; the latter, for instance, in barium titanate varies from 10 to 160 kV/cm (1 to 16 MV/m). Broadly speaking, the relative values of the three electro-mechanical constants do not vary very much, although these ratios will change when composition is altered appreciably [17].

4. Forms taken by transducers of new polycrystalline materials

This similarity of fundamental properties, embracing as it does both the extensional and rotational properties, is broadly mirrored in the physical forms which these materials assume for various applications, but inevitably the desirability of providing a means of continuous polarisation on the ferrite materials leads to some variations in form which will be illustrated later. Fig. 2 shows some of the numerous forms which barium titanate can take. Its appreciable hydrostatic piezoelectric coefficient has led to the development of many forms of microphones. Their great value lies in their robustness and high capacity, for the permittivity of this material is over 1000. In comparing such microphones made of various materials with one made from the best material in this class, lithium sulphate, corrections have to be made for circuit stray capacities and the effect of the impedance level of the device on choice of resistive loads, and consequently on the thermal noise. Making these corrections in a typical case (6 mm cube), the comparison, relative to lithium sulphate (0 dB), is: barium titanate — 7 dB, tourmaline — 16 dB, ethylene diamine tartrate — 5 dB.

From Fig. 2 well-known forms have been omitted, such as the thickness resonator, which often takes the form of a disc; the variant of the form originally described by GRÜTZMACHER in which, by curving the transducer, focussing of the radiated energy is accomplished; the tubular transducer which can radiate energy in lengthwise, radial or thickness vibration. A cheap and satisfactory alternative to the GRÜTZMACHER type is illustrated in Fig. 2C, where a toroidally shaped disc resonator has its energy concentrated back through the hole in its centre by means of a mirror.

One group of transducers depends upon the shear distortion produced by crossed fields. Some years ago MASON described [18] the prototype shown at Fig. 2F. The rectangular block was originally silvered over the faces abcd, and efgh, before the electrodes shown were applied. After polarising strongly between these original electrodes, they were dissolved off and replaced as shown, with the result

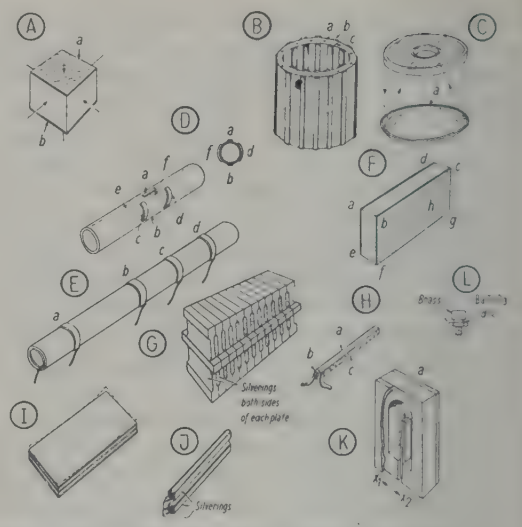


Fig. 2. Forms taken by barium titanate transducers.

A. "Hydrostatic" pressure type, silvering at a and b. Polarised from a to b.

B. Torsional type. Hollow cylinder having strip electrodes b, c etc. on inner and outer surfaces. Polarised sequentially on silverings of consecutive holes a. Shear produced, for example, by cross field b to c.

C. Toroidal type, silvering on both sides, used with concave mirror a.

D. Torsional type. a ··· f are electrodes on the outside of the tube. Polarised from c to d and from f to e. Torsional shear produced by cross field from a to b.

E. Delay line. a ··· d are electrode rings on the outside of the tube, the inner surface of which is silvered. After polarisation a ··· d to inner, a pulse is applied from a to inner and a delayed voltage pulse can be picked up on either b, c, or d to inner.

F. Shear type (see text).

G. Ventilated high intensity type.

H. Bender of flattened tube type [22]. Polarised a and c to inner silvering b.

I. Bilamellar transducer, distorts to dotted form on excitation.

J. Twister transducer (see text).

K. Directional accelerometer. Bender element a is clamped in rigid housing and responds only to motion in direction x_1—x_2.

L. Simple accelerometer (for screwing to body).

that shear in the abfe plane can occur if subsequent polarisation be applied. The resonator shown at 2D [19] illustrates one method of producing these crossed fields and a rather similar principle is illustrated at B [20] where, in effect, the inner surface of the barium titanate tube rotates on energisation with respect to the outer surface, after the tube has been polarised circumferentially using the metallising in the holes; if applied after polarising, the inner and outer coatings can be continuous. Torsional distortion can be achieved in a rather

different way as shown in Fig. 2J [21]. This is based on a simple tubular type of "bender" as at Fig. 2H [22], a web being provided between two such flattened tube elements. It is obvious that, if opposite curvatures are produced by cross-connecting the silverings after electrical polarisation, a twist in the assembly results. Such a device has also interesting possibilities for delay line applications.

The bi-lamellar structure illustrated for example at 2I, has the important difference from similar structures of the A.D.P. and rochelle salt type, that polarisation of the assembly produces curvatures of the same sign in orthogonal directions as shown. It will be clear that microphones and transmitters can be constructed in this way. The application to other types of simple bi-lamellar devices, for instance, those used in the directional accelerometer at 2K (of the type developed at Göttingen by OBERST and his co-workers [23]) and gramophone "pick-ups" [24] is obvious. A simpler type of accelerometer is shown at 2L. One other point to be emphasized in connection with Fig. 2 concerns the high internal friction of barium titanate. The mechanical Q, which SCHMIDT [25] quotes at 350, is, in our experience at N.P.L., often markedly lower in value and this leads to quite appreciable internal heating in a high intensity transducer. One way of avoiding overheating is, of course, a ventilated construction such as that shown in Fig. 2G, but it is clear that the overall efficiency can be greater for single frequency operation if the electro-mechanical coupling factor of the material is reduced, for this inevitably reduces the internal friction [26].

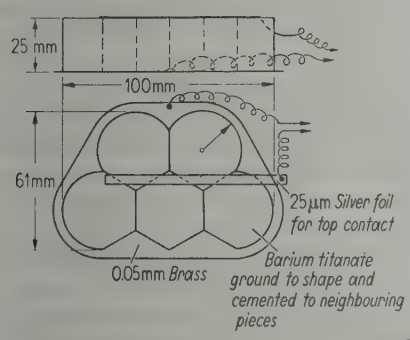


Fig. 3. Large composite barium titanate transducer for echo technique at 100 kc/s.

As a further example of the use of a material like barium titanate, Fig. 3 shows some quite large transducers [27] which were made up, using this material, to provide a method of measuring the thickness of concrete pavements and runways by an echo technique. The transducers are 100 mm across and 25 mm in thickness, so that the cost would be practically prohibitive in a material like quartz. Owing to the difficulties of making large blocks of barium titanate, it was found necessary to fabricate these transducers from five smaller pieces. MATTIAT [28] has described still larger (circa 27 cm diameter) transducers built-up in a similar way.

Fig. 4 shows the form taken by a ferrite type of torsional [29] transducer. In this case a toroidal winding is applied to the ferrite tube as shown, and to provide the crossed magnetic field necessary for shear distortion the whole ferrite tube is subjected to powerful axial magnetization. In this way shear deformation occurs, with an electro-mechanical coupling (energy $^1/_8$ basis) of 0.19 or more. Such devices are readily usable to launch torsional waves in solids, and also, of course, for viscometer work in liquids as an alternative to the A.D.P. torsional transducers used by MASON [30] and his collaborators. An alternative construction for these torsional transducers in ferrite is to employ the axial field coil for the a.c. excitation, and to polarise by means of a powerful current passed through the centre of the tube. In this case, reliance is placed on the circumferential residual magnetization of the ferrite. In favourable circumstances the performance is very good, but it is possible, by a strong longitudinal excitation, to destroy such circumferential polarisation. Excellent torsional wave delay [31] lines have, however, been made using ferrites in this way, either by using the ferrite itself as the delay medium, or by using torsional ferrite transducers with an inert propagating path in the form of a tube, rod or wire.

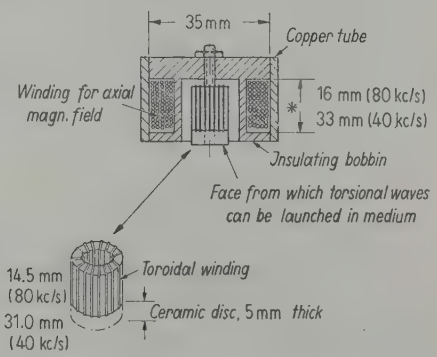


Fig. 4. Torsional ferrite transducer.

Although the mechanical Q of ferrite transducers can be very high, in the region of some thousands, its ultimate tensile strength is by no means as great as that of metals like permendur or nickel, and it seems doubtful whether it can be profitably applied to the construction of high intensity ultrasonic cutting tools which are now finding valuable application for die making and the like. A tool for such a purpose [32] made with both stator and armature entirely of *permendur* is illustrated in Fig. 5, and

tools such as these are capable of doing very rapid cutting in the hardest of materials without excessively stressing these materials. It is of interest to note that, when properly matched, the efficiency of a well-designed *permendur* transducer of this type would be over 70%.

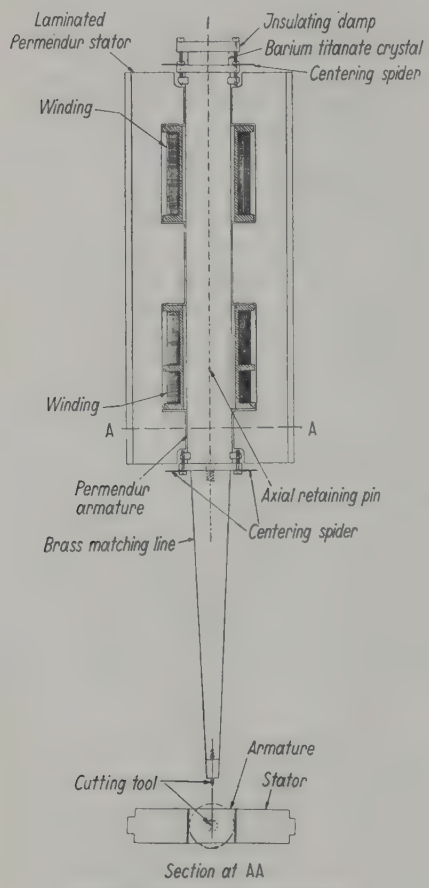


Fig. 5. Ultrasonic cutting tool.

5. A special form of transducer for pulse and wide-band operation

We will consider now the fundamental behaviour of a special transducer suitable for widespread and important applications, either with pulses (short wave trains), or for continuous wave excitation covering a very wide frequency band. A piezoelectric or a magnetostrictive material in the form of a long bar is required, and for this reason it is convenient to make it of some of the polycrystalline transducer materials considered earlier. The excitation can be either with co-linear field and mechanical displacement as in Fig. 6A and 6B or with orthogonal field and displacement as in Fig. 6H and 6I.

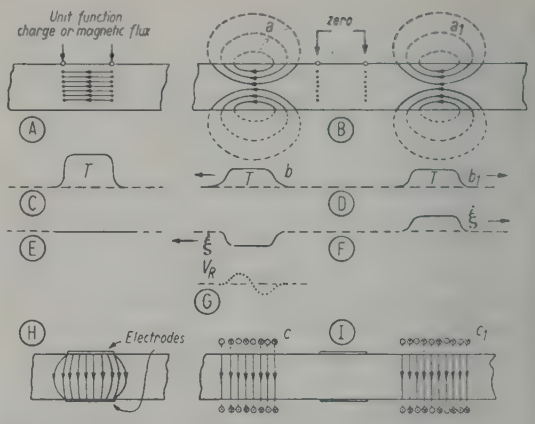


Fig. 6. "Long" type transducers:

A. "Long" type transducer with co-linear excitation and mechanical displacement. Unit function excitation of charge or of magnetic flux.

B. Conditions in A after a short time. Electric or magnetic fields a and a_1, travel to left and right. Internal field in full lines, external field in broken lines.

C. Stress Conditions in A. Ordinate is stress T. Abscissa corresponds to that of A.

D. Two stress waves b and b_1 accompanying the travelling fields in B. Abscissa corresponds to that of B.

E. Particle velocity conditions in A (i.e. zero particle velocity everywhere).

F. Particle velocity conditions in B. Two waves travel to left and right. Ordinate is particle velocity $\dot{\xi}$.

G. Modification of received electrical wave in piezoelectric case with leak between electrodes.

H. "Long" type transducer with orthogonal excitation and mechanical displacement.

J. Conditions in H after a short time. Electric or magnetic fields travel to left and right and form loops in the medium external to the bar.

In the first instance a unit function excitation, either electric or magnetic, can be applied, and there will result instantaneously a stress wave as in Fig. 6C. This, after a short time, will separate into two stress waves as at Fig. 6D, proceeding to left and right with velocity c. The corresponding particle velocity waves will first be zero as at Fig. 6E; when the waves separate they will differ in sign as at Fig. 6F. If the excitation is stopped after the wave has travelled the length l of the electrodes, a reverse pulse follows the waves illustrated. The resulting pulse pair will have a peak in its frequency spectrum at $c/2l$ where c is the propagation velocity. At the receiving set of electrodes in the piezoelectric case, as the wave passes through, a charge and voltage wave of a rather similar shape will arise in the absence of a shunt conductance. In the magnetostrictive case the current in the receiving coil will behave similarly when the resistance load on the coil is low.

It must be remembered, however, that when receiving conditions are less ideal, for instance, when appreciable shunt conductance exists, then the output electric wave will be more complex. The simple shapes at Fig. 6C and D from unit function excitation will degenerate to a shape roughly as at Fig. 6C. This will, in the case of pulse excitation of duration l/c, develop a double amplitude below the axis and be followed by an overswing.

For the sake of completeness it may be mentioned that the electrical or magnetic conditions for orthogonal field and mechanical displacement are as shown at Fig. 6H and 6I, the stress and particle velocity diagrams remaining broadly as at Fig. 6C, D, E and F. The resulting electrical wave forms which are received are not very different. This brief analysis of conditions in a locally energised bar of active material is very simple and from it can be derived, as will be seen, waveforms in various media which are well confirmed by experiment. Very similar waveforms occur in line transducers loaded with nearly matching acoustic loads on each face, provided that the joints to these loads do not act as discontinuities acoustically.

6. Waveforms of pulses in special wide-band transducers

Fig. 7 illustrates a selection of waveforms from pairs of transducers under short pulse excitation. Fig. 7a shows a waveform for longitudinal excitation with a 22 Mc/s quartz crystal launching on one side into a fused quartz bar and on the other side into a loaded plastic bar. In Fig. 7b a corresponding waveform is shown for a Y-cut 14 Mc/s quartz crystal operating in one direction on to a steel propagating path, and in the other direction into a piece of fired rock [33] (pyrophillite). The next waveform at Fig. 7c is taken from the work of ANDERSON et al. [34] and is for a rochelle salt crystal operating into air, but with the side faces of the rochelle salt damped by "Bostik" and cork so that, as a result, the mechanical energy stored in the crystal by impulsive excitation is damped out very quickly. The similarity with the foregoing waves is remarkable. CAMP [35] has described the use of barium titanate transducers so damped by high viscosity resins as to have effective mechanical Q values of about 3.

In Fig. 7e is shown the wave form from a magnetostrictive transducer using an extremely tiny coil, the nickel being in the form of a tube a $\frac{1}{4}$ mm in diameter, with a wall thickness of 0.025 mm. In Fig. 7d is shown the wave resulting from propagation from a ferrite transducer operating through an inert body in the form of a tube to excite a second similar torsional transducer. In this case the ferrite tube was 3 mm diameter. These transducer pairs constitute delay lines such as can be used in automatic computing devices.

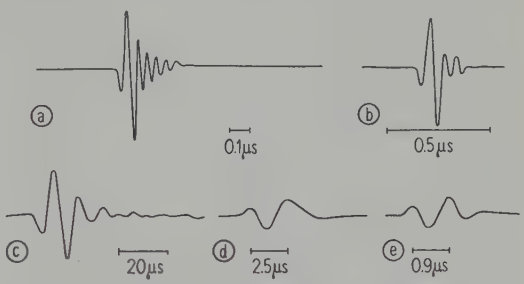


Fig. 7. Waveforms of transducers under pulse excitation:
a) 22 Mc/s X-cut quartz (heavily damped); fused quartz path,
b) 14 Mc/s Y-cut quartz (heavily damped); hard steel path,
c) 83 kc/s rochelle salt (heavily damped * sides of crystal); air path,
d) Magnetostrictive transducer; short coils on circumferentially polarised 3 mm Φ ferrite tubes; inert path, formed by 1.6 mm Φ tube; torsional waves,
e) Magnetostrictive transducer; short coils on nickel tube 0.25 mm O.D., 0.20 mm I.D.; longitudinal waves.

*) Also somewhat "snubbed" electrically [34].

Finally in Fig. 8 is shown a barium titanate transducer damped with a loaded plastic backing and operating on to a mild steel path [36]. The wave is produced from a damped electric wave train but even then is very simple and short. In this case heavy electrical damping is used which, in association with the large electro-mechanical coupling for barium titanate, enhances the mechanical damping provided. The resulting effective mechanical Q can again be as low as about 3.

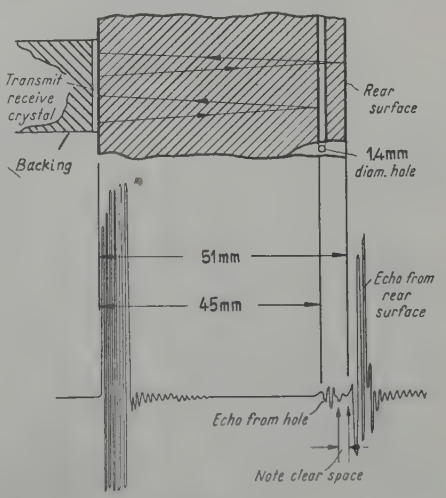


Fig. 8. Echo sounding with barium titanate; mild steel path.

7. Equivalent network analysis of special transducers

The above approach is probably not so familiar as the conventional analysis of electro-mechanical transducers involving the use of equivalent circuits, and it is of interest to see how the two agree. It is well known that on the simplest assumption electro-mechanical transducers with co-linear field and mechanical displacement yield equivalent networks of the type shown in Fig. 9a and b, where ζ_a is the impedance represented in Fig. 9c. On the other hand, on similar assumptions the second case considered above, i.e. orthogonal field and mechanical displacement, leads to an equivalent circuit as shown at Fig. 9d and e, in which the same impedance ζ_a with some multiplying factor is fed in parallel with the electrical capacitive impedance element. The situation is reversed in the magnetostrictive case. Within the limits of the simplified assumptions made in deriving these circuits, they adequately describe behaviour for wide variations in the ratios of the acoustic loadings of transducers to the characteristic impedance of the transducer itself. Over not too wide variations in these loadings it is possible to simplify the impedance to the form at Fig. 9f, and this equivalent circuit covers light symmetrical loading, slightly asymmetrical loading, and, in certain circumstances, behaviour in $\frac{1}{4}$ wave resonance. The case corresponding to the long transmission elements which we have been discussing requires some reduction in the C_m value in Fig. 9f and this simplified circuit no longer holds. The network in Fig. 9d, however, incorporating ζ_a as shown in Fig. 9c and 9f can be used, and provided that the source is of low impedance, it is clear that extremely wide band operation results. In fact, the electro-mechanical coupling affects the band-width very little, although still affecting the transfer constant. It ought also to be pointed out here that, for receiving stress waves, a piezo-electric transducer of this type on open circuit operates down to very low

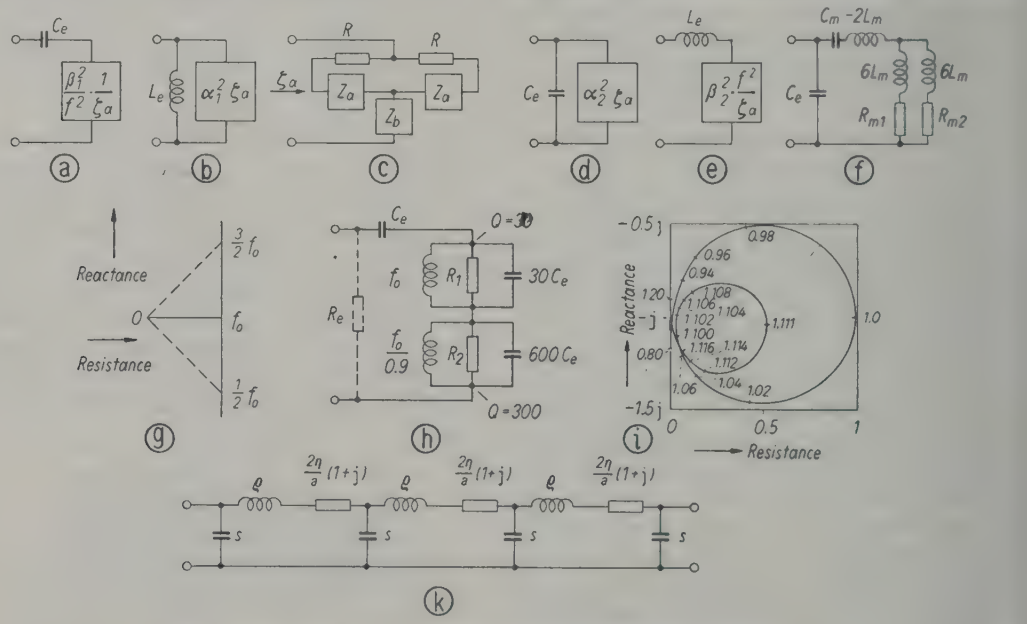


Fig. 9. Line type transducer behaviour in terms of equivalent circuits:

a) Piezoelectric: co-linear field and mechanical displacement. (f is frequency).
b) Piezomagnetic: as above.
c) Constitution of network element ζ_a; equivalent "T" of line; R_1 and R_2 = loads on faces of line transducer *).
d) Piezoelectric: orthogonal field and mechanical displacement.
e) Piezomagnetic: as above.
Note: α_1, α_2, β_1 and β_2 are frequency-invariant functions of the magnetic electric and elastic properties of the material of the transducer and of its dimensions.
f) Simplified network for transducer faces equally or unequally loaded.
g) Argand diagram for ζ_a with matched loads on both faces.
h) Complication in barium titanate equivalent network due to longitudinal-shear mode electro-mechanical coupling; radiation into water.
i) Calculated Argand diagram for equivalent circuit at h. (loss resistance R_e in h neglected).
k) Mechanical line (of radius a, density ϱ, compliance s), with surfaces damped by viscous fluid of viscosity η.

*) In the figure both indicated by R.

frequencies. Apart from this special usage, however, the band-width will be roughly from $f_0/2$ to $3f_0/2$, f_0 being the frequency $c/2l$ where c is the velocity and l the effective physical length.

It is to be noted that the use of transducers of these types generally necessitates either heavy internal damping in the path or the provision of external damping to avoid reflections from the ends. It is convenient to effect this by immersing part of the bar in grease (or in a viscous resin) and if the rigidity of the grease be neglected, the equivalent line can be regarded as damped by additional elements as shown at Fig. 9k, η being the viscosity of the grease and a the radius of the bar or line; the element $2\eta(1 + j)/a$ must not, of course, be large enough to cause appreciable reflection at the point on the bar where the damping starts.

It is probably desirable to point out here that the equivalent network for barium titanate in thickness vibration [37] is more complex than, for instance, Fig. 9f, for a shear oscillation is excited simultaneously with the extensional oscillation. For operation in water the network of Fig. 9h simulates behaviour approximately giving, for instance, the typical subsidiary peak as in Fig. 9i. The calculated frequency for this peak depends on POISSON's ratio varying from 1.14 f_0 to 1.025 f_0 as this ratio varies from 0.28 to 0.32. The peak is more pronounced when the material is partly depolarised.

A physical embodiment of the principles just described appears in Fig. 10, which illustrates one of these "long" type transducers which has been widely used for vibration studies at the N.P.L., and has enabled measurements to be carried out at temperatures ranging from — 240 °C to + 1000 °C. It comprises a coil about 8 mm long with an internal diameter of 3 mm through which a $\frac{1}{4}$ mm diameter nickel tube is threaded, small magnets being provided for polarising the stretch of tube under the coil. The mechanical impedance of the device at the free end is very small, about 800 mechanical CGS-ohms (0.8 Ns/m). The other end is extended in a grease filled tube which provides the viscosity damping

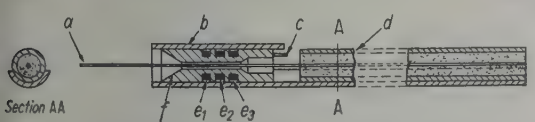


Fig. 10. Low impedance "long" type magnetostrictive transducer:
a) nickel tube, 0.25 mm O.D., 0.20 mm I.D., presents acoustic load of 800 abohms (0.8 Ns/m),
b) copper shield,
c) live terminal,
d) glass tube filled with grease,
e) sectional winding e_1, e_2, e_3 (normally all in series),
f) ivory bobbin fitted with small magnets (parallel to axis).

discussed above. It is possible to use numbers of these transducers to measure magnitudes and phases of particle velocities at various points on the surface of a body under extensional or shear vibration, and the elements attached are so fine as to interfere very little with the motion.

8. The launching process and polar diagrams of radiators on semi-infinite solids

In launching waves in or at the surface of a medium, interest is divided between the mechanical impedance of the radiator, which is needed to calculate the equivalent transducer networks, and the magnitude of the radiated field at various points in the medium. For arrays of radiators, it is further

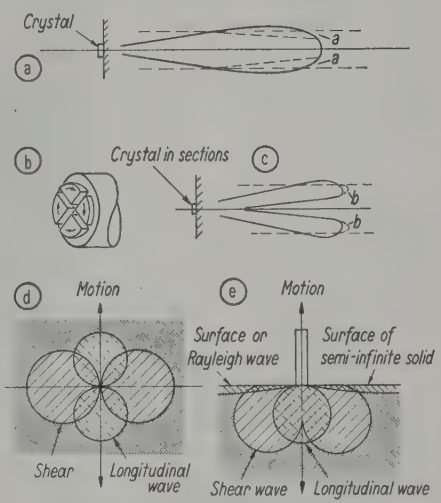


Fig. 11. Radiation into solids:
a) radiation into a rod,
b and c) torsional waves launched into a rod,
d) small radiator immersed in a solid,
e) radiator located on a semi-infinite solid.

necessary to calculate mutual impedances. While numerous investigators have dealt with the launching of waves in fluids and this subject is now well understood, only the simplest of cases for the launching of waves in solids has been tackled, for instance, dilating sources [38] and dipole sources [38] in infinite media, and large aperture radiators on semi-infinite media. When these simplifications no longer apply, the situation becomes formidably complex. A case which has been considered, at least for sinusoidal conditions, is that of a radiator applied to the end of a rod as at Fig. 11a. For a fairly extensive aperture, a narrow beam results, but when this encounters the outer boundary of the rod the longitudinal wave is reflected with change of phase and with partial mode changing. This process tends to remove energy at a and distort its distribution in the

bar, so that at one stage the wave at the centre of a rod drops to a very low intensity. In this process, of course, a considerable amount of the energy is changed to shear wave energy so slowing the propagation for sinusoidal waves although the energy can again change back to longitudinal wave energy to be picked up by a subsequent transducer further along the bar. This is the origin of the wellknown "ghost" pulse in the pulse excitation of transducers on the ends of long bars, useful to give a rough idea of shear wave velocity but embarrassing for many other reasons. It will be realised that this process, which is of course implicit in the POCHHAMMER [39] analyses, forms a fundamental limitation on the possibility of guiding or concentrating pulses of energy along long bars. It is, however, possible to concentrate the energy by means of an internal reflection at a spherical boundary surface on the solid and so avoid the difficulties which are encountered in a bar [35].

Fig. 11b shows an interesting case of propagation of a shear wave from a mosaic of Y-cut quartz crystals which can launch what is effectively a torsional wave in the form of a beam. This has a peculiar polar diagram as at Fig. 11c (to be imagined as rotated around the axis) and this time the particle velocity is tangential to the boundary surface of, for instance, a bar forming the propagation path. Such a wave is reflected without change of phase, so accentuating the polar diagram near the boundary surface as shown at b in Fig. 11c. A wave of this type can, of course, be used for delay lines [40] and also for flaw detection for surface flaws in cylindrical specimens.

While the polar diagrams of small radiators immersed in solids are relatively simple, see Fig. 11d, a much more complex situation is encountered when an attempt is made to calculate the radiation impedances and polar diagrams of radiators of small to medium aperture located on a semi-infinite medium, for three waves are radiated, the surface or RAYLEIGH wave, the shear wave and the longitudinal wave. This problem has now been solved in principle by PURSEY and MILLER [41] at the National Physical Laboratory using an integral transform technique and, although the integrals involved are very complex, many cases have now been computed. In their work they consider a) idealised radiators of linear and circular shapes, whose motion is normal to the surface, b) an idealised linear radiator moving tangentially to the surface and normally to itself, and c) an idealised circularly oscillating radiator of circular shape. Broadly speaking, the polar diagram for the normally moving radiators of small aperture is as shown at Fig. 11e while for the tangentially moving linear radiator the modes of vibration for the central and flanking lobes are interchanged. Preliminary evaluation of the shear wave polar diagrams reveals however, except in the pure torsional case, a narrow crevasse between 30° and 45° not shown on the sketches. For the torsional transducer, the energy is confined to the rotational mode and the diagram, when the aperture is small, takes the form of the lower half of the shear wave lobes of Fig. 11d.

One other aspect of the launching of elastic waves in solids from sources of extensive aperture is worthy of mention. It is often desirable to be able to direct the beam at an angle to the surface and to be able to vary that angle, and, although this can be done with the solid immersed in a liquid this brings other difficulties in its train. A much more satisfactory scheme is that shown in Fig. 12a for launching longitudinal waves in solids and at Fig. 12b for combining a mode-changing operation with the launching of the wave. In Fig. 12a optically worked plano-convex and plano-concave glass lenses were used and in Fig. 12b optically worked perspex parts were used. It will be appreciated that the possibilities of such devices in the field of ultrasonic flaw detection are immense.

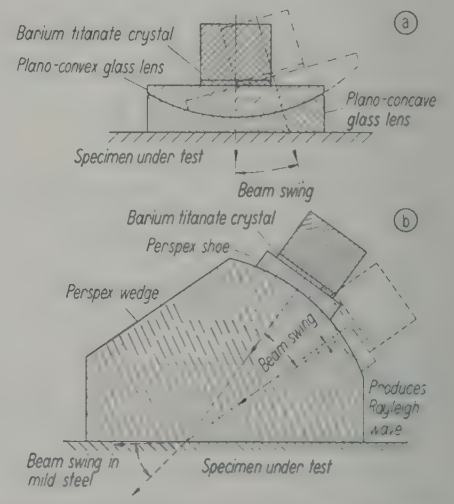


Fig. 12. Radiation into solids at oblique directions.

Finally, something further should be said about electro-mechanical transducers of the electrostatic and inductor types. BORDONI [9] and other workers at Rome have successfully used electrostatic excitation and frequency-modulation pick up for the vibration of bars down to 4 °K and this is also a standard method at the National Physical Laboratory in the metal-physics work carried out there, the devices operating up to about 0.25 Mc/s. As well-known is the inductor method used by ST. CLAIR [44] for ultrasonic precipitation of aerosols. Although his electro-acoustic transducer had an effi-

ciency of about 30 % it is doubtful if it would compete with a well-designed siren. To end this account, mention will be made of an interesting development of this inductor type of transducer shown in Fig. 13. A powerful field sets up eddy currents induced in a thin strip so as to set it in resonance longitudinally, the resonance being detected by a bridge method. This method is capable of use with non-ferromagnetic specimens as small as 20 mm × 3 mm × 0.05 mm and has also been used for the investigation of the elasticity of single crystal specimens of tiny dimensions. A variant of this device for resonating discs was successfully made at N.P.L., with the co-operation of Dr. Levi of Perugia when the latter was working there a couple of years ago.

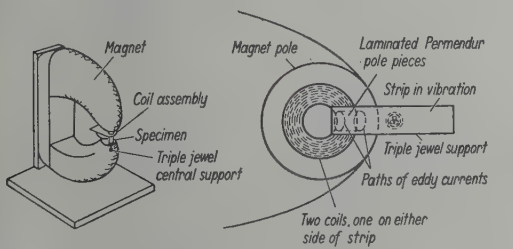


Fig. 13. Inductor type of transducer.

Some of the devices described in this account were developed as part of the research programme of the National Physical Laboratory and this paper is published with the permission of the Director of the Laboratory.

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EXPERIMENTS ON SOME ELECTRODYNAMIC ULTRASONIC VIBRATORS

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Summary. Some types of ultrasonic vibrators are described. These are made of solids of various forms which are set in vibration by the electrodynamic effect. Solids made of insulating materials are partially metallised so that they can be set in vibration. The use of such vibrators to measure elastic constants is proposed. The same vibrators may serve as ultrasonic sources for the laboratory.

Sommaire. On décrit quelques types de vibrateurs ultrasonores. Ils sont constitués par des solides de formes variées, qu'on fait vibrer par effet électrodynamique. On métallise partiellement les solides en matériau isolant pour pouvoir les faire vibrer. On propose d'employer des vibrateurs de ce genre pour mesurer les constantes élastiques. Ces mêmes vibrateurs peuvent servir de générateurs d'ultrasons dans les laboratoires.

Zusammenfassung. Es werden Ultraschallschwinger beschrieben, die aus Festkörpern hergestellt sind und durch einen elektrodynamischen Effekt angeregt werden. Auch Nichtleiter können angeregt werden, wenn man sie teilweise metallisiert. Es wird vorgeschlagen, solche Schwinger zur Messung elastischer Konstanten und als Schallquelle für Laboratoriumszwecke zu verwenden.

1. All the phenomena by which electric energy is transformed into mechanical energy are suitable for the design of ultrasonic sources, but those which have been practically constructed depend on the piezoelectric or the magnetostrictive effects.

It seemed however of interest to examine the possibility of constructing special ultrasonic vibrators depending on the mutual action of a current and a magnetic field.

The use of the electrodynamic effect in the construction of sources of sonic frequency stems from 1891, that is to say, in the electrodynamic telephone of Werner von Siemens [1]. There appeared successively, the sonic source of R. A. Fessenden [2], the loudspeaker of H. Riegger [3], those of W. Schottky [4] and E. Gerlach [5], the common cone loudspeaker of W. Rice and E. W. Kellogg [6], and finally the ribbon telephone of H. F. Olson and F. Massa [7].

All the aforementioned types of transducers, meant to operate at sonic frequency, are notably unsuitable when one passes to ultrasonic frequency because they are made of systems with lumped elements having relatively low natural frequencies.

To obtain oscillators of high frequency, it is better to make use of systems with distributed quantities and to this purpose we have realised some types of vibrator which will be described here. It might be thought that there exists a similarity between certain of the sources constructed by us and the one first described by St. Clair [8] but it should be noted that the latter is set in vibration by a particular electrodynamic effect linked with the setting up of induced currents generated by a variable magnetic field, whereas, as will be shown, the way we chose does not involve alternating magnetic fields.

2. Our experiments began with the one shown in Fig. 1. The bar shown is of glass and is metallised to render the zone shaded in the figure conducting. The exciting alternating current follows the track marked on the figure and interacts with a constant magnetic field to which one of the end faces of the bar is exposed.

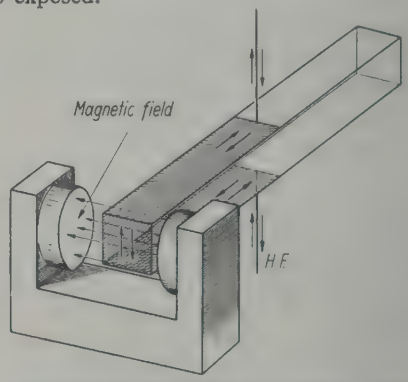


Fig. 1. Arrangement for exciting longitudinal vibrations in a bar of glass or other insulating material.

The vibromotive force of electrodynamic origin, proportional to the magnetic induction and the current intensity, has the same frequency as the latter. By suitable choice of such frequency it is possible to excite, with considerable intensity, the longitudinal vibrations corresponding to various normal modes of the bar.

This simple method suffices to set in longitudinal vibration any insulating solid at frequencies of the order of tens of kilocycles per second.

To excite torsional vibrations the excitation of the bar is arranged according to the scheme of Fig. 2.

To apply the electrodynamic method to metallic, rather than to insulating materials, it is only neces-

sary to make the exciting current transverse one end of the solid immersed in the magnetic field.

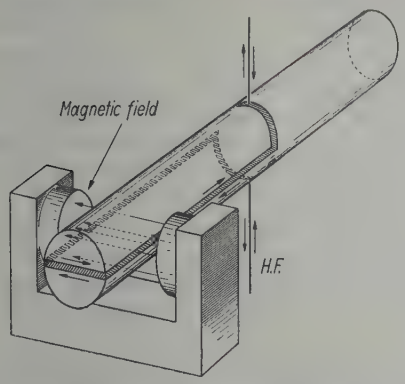


Fig. 2. Scheme of method for exciting torsional vibrations in a solid insulator.

In detail the arrangements described above can be carried out in diverse ways. Fig. 3 shows the following experimental transducers: (a) a glass bar of the type drawn in Fig. 1; (b) a bar of rubber; (c) an aluminium rod, having the ends of expanded section to allow one to be placed in the magnetic field while the other has an expanded radiating surface; (d) a transducer formed of a rectangular strip of aluminium in which the exciting current passes along an edge to which two leads are soldered; (e) a strip like the preceding, but with variable width; (f) a bar of glass on which the conducting film, sputtered on, serves to excite torsional oscillations.

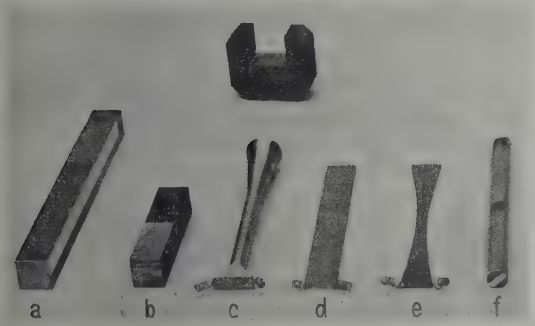


Fig. 3. Types of vibrators: (a) glass bar, (b) rubber bar, (c) aluminium bar with conical expansion, (d) aluminium strip, (e) aluminium strip, (f) glass rod for torsional vibrations.

The experimental arrangement for exciting longitudinal vibrations in a glass bar is shown in Fig. 4, whereas Fig. 5 shows how longitudinal vibrations can be excited in an aluminium strip.

The vibrators we described have been devised mainly for the purpose of measuring the elastic constants of solids. In this connection it should be remembered that several other methods founded on electrostatic, piezoelectric, magnetostrictive effects have been described in the literature [9], [10].



Fig. 4. Arrangement for exciting and revealing longitudina vibrations in a glass bar.

The electrodynamic method here proposed seems more interesting for longitudinal vibrations of highly damped solids because of the comparatively high forces obtainable by electrodynamic action with respect to those attainable by electrostatic methods.



Fig. 5. Experimental arrangement for exciting longitudinal vibrations in a metallic strip furnished with a trumpet for irradiating the air with ultrasonics.

It should be added that the method of exciting torsional vibrations illustrated in Fig. 2 seems to have special interest because the present methods (including the electrostatic) do not favour torsional oscillations in cylindrical rods whereas the coupling of other vibrating systems (like the piezoelectric quartz, ferromagnetic strip, etc.) to the bar by making torsional vibration possible, requires the introduction of corrections to the results *).

3. Though the scope of our research was mainly concerned with a new method for determining elastic constants in solids through velocity measure-

*) As an exception one might quote the method described during this Congress by G. BRADFIELD.

ments, we wish to point out that each of the systems considered — excepting those intended for torsional oscillations — forms a source apt to irradiate the ambient medium, if supplied with adequate electric power and a matched radiating system. For example, on Fig. 5, is visible a trumpet coupled to a metallic vibrating strip which results in a greater output in the transfer of acoustic power into vibration of the ambient medium.

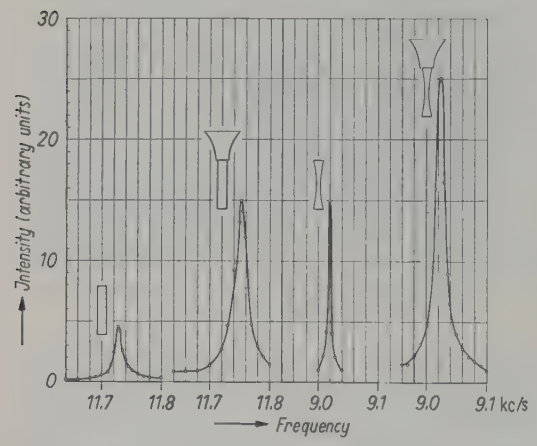


Fig. 6. Response curves of sources formed of aluminium strips in longitudinal vibration, with and without radiation trumpet. The first pair of curves refer to source d of Fig. 3, the second pair to source e.

In connection with these last considerations, response curves relative to the emission from the strip of aluminium with and without trumpet have been traced. For reasons of convenience in the use of the apparatus, a frequency above the audible limit is selected; Fig. 6 shows the resulting curves. One observes that the use of the trumpet (see the schematical drawings of the vibrators near the curves), besides giving the emission a directional character, puts greater energy into the medium, as shown by the diminished Q factor.

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GÉNÉRALISATION D'UNE MÉTHODE ÉLECTROSTATIQUE POUR LA MESURE ULTRASONORE DES CONSTANTES ÉLASTIQUES ET ANÉLASTIQUES DES SOLIDES

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Sommaire. Exposé des perfectionnements apportés à une méthode électrostatique de mesure des constantes élastiques et anélastiques des solides. On a utilisé des vibrations de traction, de torsion et autres modes de déformations dont la fréquence atteint jusqu'à 300 kHz.

Zusammenfassung. Es wird eine verbesserte elektrostatische Meßmethode beschrieben, mit der man elastische und andere Konstanten von Festkörpern mit Hilfe von Dehn- und Torsionswellen bis zu Frequenzen von 300 kHz bestimmen kann.

Summary. Improvements are described in an electrostatic method for measuring the elastic and anelastic constants of solids by means of longitudinal and torsional vibrations up to frequencies of 300 kc/s.

1. Introduction

La détermination des constantes élastiques et anélastiques d'un solide s'effectue assez souvent à l'heure actuelle par une méthode dynamique, en mesurant la vitesse de propagation et l'atténuation des ondes ultrasonores, et en utilisant les relations bien connues qui lient ces grandeurs aux paramètres élastiques.

Pour exciter les vibrations et pour mesurer leur amplitude on peut utiliser tous les phénomènes qui donnent lieu à une transformation d'énergie électrique en énergie mécanique ou à la transformation inverse. On a donc les méthodes de mesure piézoélectrique [1] ··· [5], ferromagnétiques [6], électrodynamiques [7] ··· [9], magnétostrictives [10] et électrostatiques [11] ··· [13].

Ces dernières ont reçu jusqu'à présent des applications particulièrement nombreuses en raison des avantages suivants:

a) elles peuvent être employées entre des limites étendues de températures depuis les plus basses valeurs réalisables jusqu'au point de fusion;

b) elles donnent lieu à des efforts mécaniques très petits, qui rendent négligeables les effets de second ordre dans les déformations élastiques et anélastiques;

c) elles n'augmentent pas de façon appréciable la dissipation élastique du matériau, contrairement à ce qui se passe pour la plupart des méthodes utilisant des champs magnétiques ou des couplages mécaniques entre le matériau et l'appareil de mesure.

Dans ce qui suit, on exposera en détail une méthode électrostatique qui permet d'exciter les vibrations et de mesurer leurs amplitudes à l'aide d'une seule électrode.

Cette méthode a été employée dans plusieurs travaux sur les vibrations de traction et de flexion jusqu'à des fréquences de 50 kHz [14] ··· [20].

Elle a été récemment perfectionnée de façon notable en élevant la limite des *fréquences* de vibrations mesurables puis le nombre de *types des vibrations* utilisables. On a pu ainsi mesurer jusqu'à des fréquences de 200 à 300 kHz et l'on a pu également exciter et mesurer avec une seule électrode et sur la même éprouvette soit des *vibrations de torsion*, soit des *vibrations de traction*. Il est alors aisé de mesurer le coefficient de Poisson et de suivre ses variations en fonction de la température.

On a pu également exciter et mesurer les vibrations de solides de forme *sphérique ou annulaire*. Par comparaison avec les fréquences de résonance relatives aux différents types de vibration, on arrive à déterminer toutes les constantes élastiques des différents solides et on peut contrôler leur isotropie et leur homogénéité.

2. Mesures en haute fréquence

La méthode électrostatique en haute fréquence a été décrite en détail dans une communication précédente [12]. Il suffit donc de rappeler que les vibrations sont excitées par l'attraction électrostatique due à une tension alternative appliquée entre le solide et une électrode très voisine. La même électrode sert aussi à la mesure de l'amplitude des vibrations. Celles-ci produisent en effet une variation de la distance et par conséquent de la capacité entre l'électrode et le solide. Ainsi on peut obtenir une modulation de fréquence dans un circuit à haute fréquence dont la capacité est constituée par le solide et l'électrode. La modulation de fréquence est transformée en modulation d'amplitude à l'aide du circuit convertisseur proposé par RIEGGER [21]. Elle est ensuite démodulée. On obtient ainsi une tension alternative de même fréquence que les vibrations, et dont la valeur est proportionnelle à leur amplitude.

Pour mesurer les vibrations jusqu'à des fréquences de 200 à 300 kHz on a, en premier lieu, réalisé des amplificateurs dont la réponse est constante jusqu'à ces fréquences.

Dans le même but on a réalisé un convertisseur de modulation assez sensible aux hautes fréquences, pour compenser la réduction de la tension de sortie produite par la diminution de l'amplitude des vibrations. En effet pour une éprouvette donnée et une valeur donnée du coefficient de résonance et de la force vibromotrice, cette amplitude est inversement proportionnelle au carré de la fréquence. Pour la haute fréquence on a donc choisi une valeur moyenne à peu près égale au produit de la plus haute fréquence de vibration à mesurer (200 à 300 kHz) par le coefficient de résonance du circuit convertisseur (de l'ordre de 50 ··· 100).

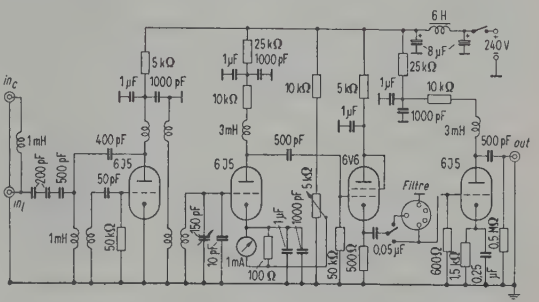


Fig. 1. Schéma du vibromètre.

Un tel dispositif (Fig. 1) a permis, par exemple, de mesurer les valeurs du coefficient de résonance d'une éprouvette d'aluminium (99,5%) jusqu'à la fréquence de 250 kHz (Fig. 2). Les mesures ont été effectuées sous vide, car la viscosité de l'air aurait produit une dissipation beaucoup plus grande que celle qui caractérise l'éprouvette. Il a fallu aussi réduire au minimum la dissipation due à la fixation de l'éprouvette. Celle-ci était soutenue par trois pointes d'acier (1 mm de diamètre) situées dans un plan nodal. On a vérifié que l'effet des pointes était vraiment négligeable quand on les écartait du plan nodal de façon à augmenter le couplage mécanique entre l'éprouvette et les organes de fixation; on

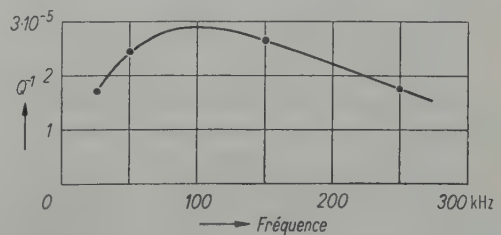


Fig. 2. Dissipation anélastique dans une éprouvette d'aluminium (99,5%) en fonction de la fréquence.

observait alors que ce déplacement augmente très peu la dissipation.

Lorsqu'il suffit de mesurer les constantes élastiques, on a remarqué qu'on peut simplifier beaucoup le dispositif expérimental en appuyant directement l'éprouvette sur l'électrode, par l'intermédiaire d'une feuille très mince en cellophane ou d'un autre diélectrique, ce qu'avait déjà fait IDE [22] pour l'excitation des vibrations. On a remarqué que les petites ondulations du diélectrique empêchent l'éprouvette de toucher l'électrode sur toute sa surface et donnant lieu à un couplage mécanique très faible, et à des altérations des fréquences de résonance qui, en général, ne dépassent pas 1%.

3. Vibrations de torsion et mesure du coefficient de Poisson

On a pu aussi employer la méthode électrostatique pour étudier les vibrations de torsion. Dans ce cas toutefois il faut employer des éprouvettes prismatiques. Pour produire le couple de torsion, l'électrode doit être placée dissymétriquement par rapport à un plan passant par l'axe de l'éprouvette tout en étant normal à une paire de faces. L'éprouvette est supportée par des pointes disposées dans un plan nodal (Fig. 3) ou bien elle s'appuie directement sur une électrode dissymétrique de forme convenable, avec interposition d'une feuille de substance diélectrique. Dans cette dernière disposition l'électrode étant placée près de la section moyenne de l'éprouvette, on peut exciter aussi les vibrations de traction en se servant de la contraction latérale du matériau. On peut donc mesurer en même temps le module de YOUNG et le module de torsion et suivre sans difficulté les variations de ces paramètres et du coefficient de POISSON en fonction de la température.



Fig. 3. Support pour l'excitation et la mesure des vibrations de torsion. E: électrode.

Comme le solide n'est soumis à aucun effort, on peut même pousser les mesures jusqu'aux environs du point de fusion et obtenir le coefficient de Poisson avec une précision satisfaisante.

En ce qui concerne les vibrations de torsion, le Tableau I donne les valeurs des 15 premières fréquences de résonance mesurées sur une éprouvette d'aluminium à section carrée de 200 mm de long. On voit que les rapports entre les fréquences sont, avec une très bonne approximation, des nombres entiers, ce qui démontre à la fois l'isotropie du matériau et la bonne approximation de la théorie mathématique du premier ordre.

Tableau I: Spectre des fréquences correspondant aux modes de torsion d'une barre en aluminium à section carrée, 11,95 mm × 11,95 mm × 200 mm.

Ordre du mode de vibration	Fréquence Hz
1	7,096
3	21,270
5	35,446
7	49,550
9	63,702
11	77,897
13	92,031
15	106,170

4. Vibrations des sphères et des anneaux

Pour exciter et mesurer les vibrations d'une sphère il suffit d'appuyer celle-ci sur l'électrode avec interposition d'une mince couche de matériau diélectrique. On peut aussi appuyer l'électrode sur la sphère (Fig. 4); dans les deux cas il est facile de distinguer les différents types de vibration, comme on l'a fait dans le Tableau II, pour les premiers modes d'une sphère d'aluminium de 100 mm de diamètre.



Fig. 4. Dispositif expérimental pour l'excitation et la mesure des vibrations d'une sphère et d'un anneau. E: électrode.

La figure 4 montre également le dispositif expérimental qui permet d'exciter les vibrations d'un anneau. Ce dernier est soutenu par des fils, et une petite électrode est appuyée sur sa partie supérieure avec interposition d'une feuille de matériau diélectrique. Avec ce dispositif on se borne à exciter et à mesurer les vibrations de traction ou de flexion ayant lieu dans le plan même de l'anneau. L'accord entre les fréquences calculées et les données expérimentales est particulièrement satisfaisant même pour

Tableau II: Fréquences de résonance d'une sphère en aluminium.
D = 100 mm

No	Fréquence de résonance en Hz	Type de la vibration
1	26,591	1° Sphéroidale
2	39,450	—
3	50,729	1° Radiale
4	55,967	1° Rotatoire
5	71,360	—
6	71,432	—
7	81,288	—
8	107,156	—
9	110,612	—

Tableau III: Fréquences de résonance d'un anneau en aluminium commercial ayant les dimensions suivantes:
diamètre extérieur 100 mm
diamètre intérieur 70 mm
épaisseur 8 mm

Fréquence de résonance en kHz	Mode de vibration
5,2	1° Mode flexion
7,0	2° „ „
13,23	3° „ „
19,61	1° „ extension
23,17	4° „ flexion
26,84	2° „ extension
33,93	5° „ flexion
41,68	3° „ extension
48,40	6° „ flexion
57,01	4° „ extension
68,60	7° „ flexion
75,00	5° „ extension
80,55	8° „ flexion
92,20	6° „ extension
103,7	9° „ flexion
115,1	7° „ extension
124,9	10° „ flexion
134,2	8° „ extension
147,4	11° „ flexion
153,15	9° „ extension

les modes d'ordre élevé. On peut alors distinguer aisément les différents types de vibration comme il est indiqué sur le Tableau III, et mesurer la dissipation pour un même type de vibration dans un intervalle trés étendu de fréquence.

5. Conclusions

Les perfectionnements récemment apportés à la méthode de mesure électrostatique en haute fréquence permettent de mesurer la vitesse et l'absorption des ultrasons jusqu'à des fréquences de 200 à 300 kHz pour presque tous les types de vibrations.

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VISUALISATION DES ONDES SONORES ET ULTRA-SONORES PAR LA MÉTHODE DES STRIATIONS *)

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Sommaire. La méthode des striations a permis de voir directement dans l'air des ondes sonores de fréquence 4000 Hz. En les faisant traverser certains produits (laine de verre par exemple) ou en les faisant se réfléchir sur eux, on peut juger directement par photographie de l'importance de leurs facteurs de transmission et de réflexion. On peut juger de même de l'efficacité de certaines surfaces (plafonds comportant différents reliefs: poutrelles, caissons etc.) au point de vue de leur pouvoir réfléchissant, absorbant et diffusant. L'emploi des ultra-sons sur des maquettes réalisées avec une réduction appropriée permet immédiatement de se rendre compte du rôle de la nature des parois. Enfin la réflexion des ultra-sons dans l'air sur une surface plane, observée soit stroboscopiquement

*) En vue de l'impossibilité de réduire l'étendue du texte suffisamment pour permettre sa publication dans ce Compte Rendu, seuls les trois sommaires sont inclus ici. On trouvera le texte et la discussion dans Acustica **4** [1954], No. 2.

soit en lumiere continue, permet de se rendre compte du mécanisme de la formation des ondes stationnaires parallèles à la surface et de déterminer une vitesse de phase plus grande que la nature du son: on obtient ainsi un excellent schéma de la propagation des ondes électromagnétiques dans un tuyau.

Zusammenfassung. Mit Hilfe einer Schlierenmethode ist es möglich, Schallwellen mit einer Frequenz von 4000 Hz in Luft unmittelbar sichtbar zu machen. Läßt man die Wellen durch gewisse Materialien (z.B. Glaswolle) hindurchlaufen oder daran reflektieren, so kann man sich nach einer photographischen Aufnahme unmittelbar ein Urteil über die Ausbreitungs- und Reflexionskoeffizienten bilden. Ebenso kann man die Wirksamkeit verschiedener Oberflächen (Decken mit kasten- oder balkenförmigen Strukturen usw.) hinsichtlich ihrer Reflexions-, Absorptions- und Streuungseigenschaften beurteilen. Bei Anwendung der Methode auf die Untersuchung von Modellen in passender Verkleinerung kann man sich sofort über die Auswirkung der Beschaffenheit der Wände ein Bild machen. Schließlich gibt die Ultraschall-Reflexion in Luft an einer ebenen Fläche, wobei man sowohl stroboskopische als auch ununterbrochene Beleuchtung verwenden kann, Aufschluß über die Ausbildung von stationären Wellen parallel zur Oberfläche und über die dabei auftretende erhöhte Phasengeschwindigkeit. Auf diese Weise erhält man ein ausgezeichnetes Analogieschema für die Ausbreitung elektromagnetischer Wellen in einem Rohr.

Summary. The method of striations can be used to observe directly sound waves of frequency 4000 c/s. By making the waves traverse certain products, e.g. glass wool, or by reflecting them from their surfaces, one can judge from photographs the relative importance of their transmission and reflection factors. One can even judge the efficacy of certain surfaces (ceilings of different relief forms like caissons and girders) from the point of view of their reflecting, absorbing and diffusing powers. The use of ultrasonic waves on models at appropriate scale permits the calculation of the role of the nature of the walls in an enclosure. Finally, reflexion of ultrasonics in air from a plane surface, observed either stroboscopically or under continuous illumination, can be used to investigate the formation of stationary waves parallel to the surface and to determine phase velocities greater than normal; so an excellent picture of the propagation of electromagnetic waves in a tube can be obtained.

EXPERIMENTS ON A TUNED TRANSDUCER FOR THE ULTRASONIC RANGE

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Summary. Experiments on ultrasonic transducers formed of a piezoelectric element of barium titanate between two rods of aluminium are described. These transducers, used as microphones in the air, have a good sensitivity (70 mV/dNm^{-2}) and a Q of 2500. The same transducers can be used as ultrasonic sources of comparatively high intensity.

Sommaire. On décrit des essais effectués sur des transducteurs ultrasonores constitués par un élément piézoélectrique de titanate de baryum placé entre deux tiges d'aluminium. Ces transducteurs, employés comme microphones dans l'air, ont une bonne sensibilité (70 mV/dNm^{-2}) et un Q égal à 2500. On peut les employer comme générateurs d'ultrasons relativement intenses.

Zusammenfassung. Es werden Experimente mit Ultraschallwandlern beschrieben, die aus einem zwischen zwei Aluminiumstäben angebrachten Element aus Bariumtitanat bestehen. Diese Wandler haben als Mikrophone in Luft eine recht gute Empfindlichkeit (70 mV/dNm^{-2}) bei einem Gütefaktor von 2500. Dieselben Wandler können auch als Schallquelle von relativ hoher Intensität verwendet werden.

1. Microphones intended for use at sonic frequencies have usually been designed ideally to have constant sensitivity to varying frequency. When, however, it is a question of picking up ultrasonic waves, one may wish to receive a single frequency and to get the maximum sensitivity at this frequency. Uniformity of response may even be disadvantageous from the point of view of getting a good signal to noise ratio.

In the technique of underwater signalling tuned microphones made up of piezo-electric [1] or magneto-strictive [2] sandwiches have been used, these microphones being sensitive only to frequencies which correspond to the normal modes of such systems.

Concerning microphones destined for use in the air, we may mention a type made of a nickel rod, used by SLAYMARE and MEEKER [3] to detect the presence of obstacles to the blind.

In this connection we thought it would be of interest to construct some selective microphones to pick up in the air ultrasonic signals having a frequency in the range 10 kHz to 50 kHz.

2. One of the microphones is illustrated schematically in Fig. 1. The element converting mechanical

into electrical energy is made of a ceramic disc of barium titanate (24 mm diameter; 2.7 mm thick) which normally resonates at 1 Mc/s. To this are attached, through adhesive joints, two rods of aluminium 200 mm long. The whole system then resonates at a fundamental of 6.35 kc/s having a pressure anti-node at the centre and nodes at the ends. It can vibrate in all the harmonics of the given frequency, although only those of odd order can be used in a microphone, since it is obviously necessary that the piezoelectric element should be at a pressure anti-node.

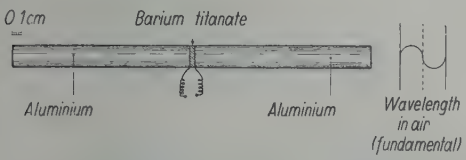


Fig. 1. Scheme of microphone a; frequencies 6.35, 19, 32, 44, 57 and 70 kc/s.

Fig. 2a shows the microphone described above; the one shown in Fig. 2b has a fundamental of 16.2 kc/s while that of Fig. 2c has a fundamental of 5.4 kc/s; the latter is furnished with a cone at one end to increase the sensitivity.

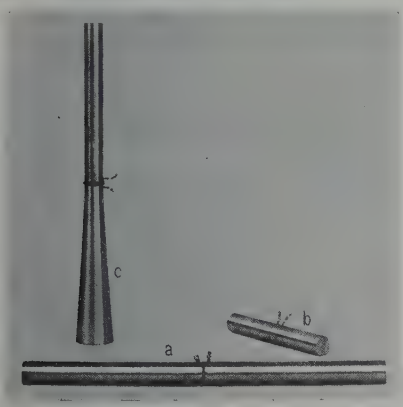


Fig. 2. Photograph of three microphones.

In order to study the operating characteristics of the microphone independently of the structure of the sound field, the following procedure was carried out. A plane electrode, to which was led an alternating potential, was put as near as possible to one of the terminal faces of the microphone. In this way a sinusoidal vibromotive force, of constant amplitude and variable frequency, was applied, the force being evidently double the frequency of the alternating potential.

Fig. 3 shows the response curve so obtained for microphone of Fig. 2a. It is apparent that the microphone has extremely selective characteristics, and a rapid calculation shows that the Q factor, for the fundamental, is of the order of 2500.

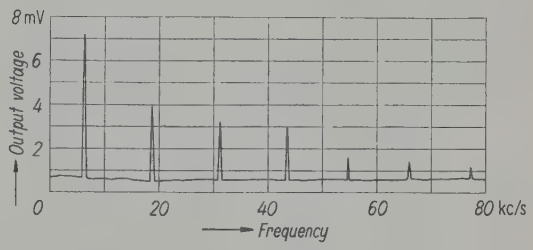


Fig. 3. Response curve of microphone a; abscissae, frequency of vibromotive force, ordinates, potential difference at barium titanate element.

Some measurements were also carried out in free field, with plane waves incident normally on the terminal faces of the microphone, resulting in the following values of the sensitivity at the fundamental:

microphone a: sensitivity 21.5 mV/dNm^{-2}
microphone c: sensitivity 74 mV/dNm^{-2}.

Such values, besides demonstrating the high sensitivity of the microphones, show the advantage of the coned end for increasing the sensitivity by facilitating the transfer of energy between sound field and transducer.

3. The transducers previously described may be inverted, and can then work as loudspeakers. Some preliminary experiments have shown a good performance in these conditions and a satisfactory emission of sonic energy has been observed. For instance the sound intensity at 1 meter distance from transducer c, supplied with 50 V, at its fundamental frequency, is 110 dB.

4. From these experiments, and others in progress, we can infer that the above described transducers can be used either as receivers in ultrasonic signalling at short distances in the air or as powerful ultrasonic sources.

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AKUSTISCHE LEISTUNGSBESTIMMUNG WÄHREND DER THERAPEUTISCHEN BEHANDLUNG MIT ULTRASCHALL-GERÄTEN

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Zusammenfassung. Der Strahlungswiderstand von einem zur Wellenlänge großen Schwinger ist durch das Produkt aus Wellenwiderstand des Mediums und Abstrahlfläche gegeben. Während der Wellenwiderstand des menschlichen Körpers vom Schwinger aus konstant und dem Wasser gleich erscheint, kann sich die strahlende Fläche bei der therapeutischen Behandlung ändern, wenn der Schwinger nicht vollkommen den Körper berührt. Ein Therapiegerät sollte diesen Möglichkeiten begegnen. Eine zweckmäßige Lösung umfaßt die laufende Messung des Strahlungswiderstandes während der Behandlung. Von den piezoelektrischen Wandlern eignet sich Lithiumsulfat für Therapieschwinger am besten, da die Anpassungsverhältnisse hier besonders günstig sind. Die bei mangelhafter Flächenkopplung auftretenden Änderungen des Wandlerklemmenwiderstandes und der abgestrahlten Leistung werden angegeben. Die Abstrahlung vorgewählter Leistungen ist nur möglich, wenn man die Schwingerwiderstandsänderung dazu benutzt, den Therapeuten darauf aufmerksam zu machen, daß er für hinreichende Flächenberührung zu sorgen hat.

Summary. Whereas the wave-resistance of the human body when irradiated is constant and equal to that of water the effective radiating area of a transducer (at best lithium sulphate) in ultrasonic therapy may change during the treatment, so that its radiation resistance ought to be measured while it is in operation. When this is done, changes in this factor can be used to warn the therapeutist to take care that the radiating surface under the clamp has not changed.

Sommaire. La résistance de rayonnement d'un vibrateur de grandes dimensions par rapport à la longueur d'onde est égale au produit de la résistance d'onde du milieu par la surface de rayonnement. Alors que, à partir du vibrateur, la résistance d'onde du corps humain paraît constante et égale à celle de l'eau, la surface rayonnante peut varier pendant le traitement thérapeutique, si le vibrateur n'est pas en contact parfait avec le corps. D'ou la nécessité de mesurer en permanence la résistance de rayonnement pendant le traitement. Le sulfate de lithium s'est révélé le meilleur des cristaux piézoélectriques pour vibrateur d'appareil de thérapie et il est indispensable d'établir un contact satisfaisant des surfaces.

Bei der therapeutischen Einstrahlung von Ultraschall in den menschlichen Körper ist die Messung und fortlaufende Einhaltung einer vorgewählten akustischen Leistung erwünscht, um dem Arzt die Möglichkeit einer reproduzierbaren Dosierung geben zu können. Die US-Energie wird von einem elektroakustischen Wandler, meistens vom piezoelektrischen Typ, an das Medium abgegeben. Ein Maß für die abgestrahlte Schalleistung ist der akustische Strahlungswiderstand. Er ist bekanntlich für einen Wandler, dessen effektiv strahlende Fläche S groß zur Wellenlänge ist, durch

$$r_s = \varrho c S$$

gegeben. c und ϱ sind Schallgeschwindigkeit und Dichte des bestrahlten Mediums. Er hängt also sowohl vom Wellenwiderstand ϱc des Mediums als auch von der strahlenden Fläche des Wandlers ab.

Das hier vorliegende Medium, der menschliche Körper, ist jedoch weder geometrisch als unendlich ausgedehnt, noch durch die eingelagerten Knochen als reflexionsfrei zu betrachten; Bedingungen, an die die Definition des Wellenwiderstandes geknüpft ist. Allerdings ist die Schallschwächung bei der in der US-Therapie benutzten Frequenz von ca. 1 MHz bereits relativ groß (Halbwertsschicht im Muskel ca. 3 cm), so daß eine Abweichung des Wellenwiderstandes nur bei Extremitäten kleiner Lineardimensionen und bei kleinen Abständen des Knochens von der Hautoberfläche zustande kommen könnte.

Eine Untersuchung, welche Werte sich für den Wellenwiderstand von der Wandlerseite bei konstanter Strahlerfläche ergeben, zeigt jedoch, daß er für den um 1 MHz liegenden Frequenzbereich an allen Stellen des menschlichen Körpers von der Haut aus gesehen praktisch konstant und dem des Wassers gleich ist [1], d.h., der infolge Reflexion an der Rückseite von Extremitäten von der Einstrahlseite aus gesehen und am Knochen bedingte Energierückwurf auf den Wandler ist bei den üblichen Schwingerabmessungen (Durchmesser etwa 10 bis 20 Wellenlängen) entweder sehr klein oder erfolgt bei geringen Linearausdehnungen oder kleinen Abständen des Knochens von der Hautoberfläche in solchen Phasenlagen, daß er sich nicht auf die Abstrahlung des Wandlers auswirken kann. Damit ist prinzipiell eine Messung und Eichung der abgestrahlten Leistung in Wasser möglich, die dann mit genügender Genauigkeit der in den Körper eingestrahlten Energie entspricht.

Bei der üblichen Behandlungstechnik wird der US-Schwinger massierend über einen bestimmten Körperbezirk bewegt. Wenn man bei dieser Applikationsart nicht sorgfältig arbeitet, d.h. eine ungenügende Menge oder Art von Koppelsubstanz be-

nutzt oder den Schwinger über unebene Körperstellen bewegt, kann der Wandler mit seiner Abstrahlfläche den Körper unvollkommen berühren, so daß die eingestrahlte Leistung keineswegs mit der anfänglich vorgewählten übereinstimmt.

Die vom akustischen Standpunkt selbstverständliche Forderung, den Wandler bei der Behandlung immer in vollständigen Kontakt mit dem Körper zu bringen, wird vom Arzt häufig nicht genügend beachtet. Er kann deshalb zu unkontrollierbaren therapeutischen Effekten kommen.

Die Forderung an ein Therapiegerät ist also, diesen mangelhaften Flächenberührungen des Therapiekopfes mit dem Patientenkörper zu begegnen. Das müßte sich am zweckmäßigsten durch eine laufende Messung des Strahlungswiderstandes während der Behandlung erreichen lassen.

Bei der Auswahl von für dieses Frequenzgebiet allein in Frage kommenden piezoelektrischen Wandlern stehen die Materialien Quarz, Bariumtitanat, Ammoniumdihydrogenphosphat (ADP) und Lithiumsulfat (LSH) zur Auswahl. Während sich Quarz seit langer Zeit bewährt hat, haftet ihm doch der Nachteil der kleinen Piezokonstanten (X-Schnitt mit $e_{11} = 0{,}16$ As/m²) an. Für die Erregung des Wandlers benötigt man für eine Intensität von 3 W/cm² in Wasser etwa 2,7 kV, so daß ein relativ starkes und damit unbewegliches Kabel benutzt werden muß, um die hohe Spannung dem Schwinger zuzuführen. Zwar besteht die Möglichkeit, eine Spannungstransformation im Handgriff des Therapiekopfes vorzunehmen, um das Zuführungskabel mit kleinerer Betriebsspannung zu belasten, doch tritt eine unerwünschte Erwärmung durch die verlustbehaftete Transformation auf. Bariumtitanat (polarisiert) mit einer Piezokonstante von etwa 10 As/m² hat den Vorteil, daß man für die gleiche Intensität mit einer erregenden Spannung von weniger als 100 V auskommt, jedoch sind die inneren Verluste im Wandler groß, so daß Strahlungswiderstandsänderungen für den vorgesehenen Zweck nicht genau genug gemessen werden können. ADP kommt nicht in Frage, da es sich bis zu Intensitäten von 3 W/cm² nicht betriebssicher erregen läßt. Mit LSH hat man ein Material, das mit den geringen Werten seiner elektrischen und mechanischen Verluste dem Quarz am nächsten kommt, so daß die Strahlungswiderstandsänderungen bei verschiedener Berührung der strahlenden Fläche hinreichend bestimmt werden können. Die Piezokonstante für den Y-Schnitt beträgt etwa 0,85 As/m². Die benötigte erregende Spannung für 3 W/cm² liegt bei etwa 800 V, so daß hier noch ein relativ dünnes Zuführungskabel benutzt werden kann.

Bei einem Therapiekopf wird der LSH-Kristall auf eine Metallplatte gekittet. Die Abstrahlung von Querkontraktionen wird hierbei weitgehend unterbunden, so daß nur die Longitudinalschwingung des Wandlers übertragen wird. Es resultiert für das gekoppelte System eine definierte Resonanzstelle. Die Leistungsabgabe wird dann optimal, wenn bei Abstrahlung von der Metallplatte in Wasser und Angrenzung der rückwärtigen Kristalloberfläche an Luft ein Druckknoten in die Kittschicht verlegt wird. Durch bestimmte Dimensionierung der Metallplatte und des Kristalldurchmessers läßt sich eine Amplitudenverteilung auf der Abstrahlfläche erreichen, die etwa glockenförmig ist. Sie hat eine Schalldruckverteilung im Nahfeld zur Folge, die auf der Mittelachse des abgestrahlten Schallbündels um etwa $\pm$ 2,5 dB schwankt (Abb. 1a). Ebenso ausgeglichen sind die Querverteilungen im Nahfeld (Abb. 1b).

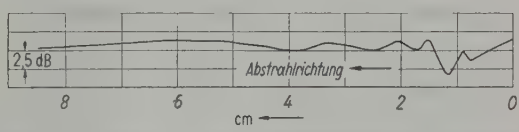


Abb. 1a. Nahfeld eines Therapieschwingers auf der Mittelachse des abgestrahlten Schallbündels.

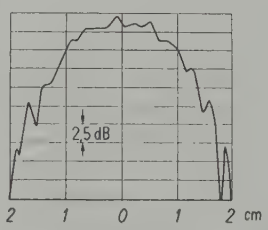


Abb. 1b. Querverteilung im Nahfeld unmittelbar vor dem Therapieschwinger.

Die sich in Luft (2) und Wasser (1) aus Brückenmessungen ergebenden Ortskurven des relativen Scheinleitwertes eines solchen Therapieschwingers mit Kabel sind in Abb. 2 wiedergegeben. Aus ihnen läßt sich der Gesamtwirkungsgrad nach bekannten Verfahren berechnen, der für diesen Fall bis 3 W/cm² etwa 60% beträgt. Die effektiv strahlende Fläche, die zur Angabe der abgestrahlten Intensität erforderlich ist, ergibt sich aus der Richtkennlinienbreite im Fernfeld hier zu etwa 7 cm².

Eine unmittelbare, laufende Überwachung des Strahlungswiderstandes und damit der effektiv an den Körper angrenzenden Fläche des Therapiekopfes während der Behandlung ist nicht ohne weiteres möglich. Denkt man sich nämlich den Wandler aus zwei Teilen aufgebaut, von denen nur ein Teil δ an das zu behandelnde Medium, der andere (1-δ) an Luft grenzt, so liegen im bekannten elektrischen Ersatzbild beide Wandlerteile parallel (Abb. 3). Hier bedeutet R_v und R_m den elektrischen und mechanischen Verlustwiderstand, R_w den elek-

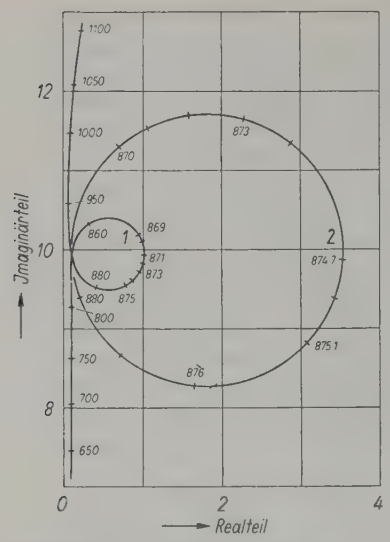


Abb. 2. Ortskurven des Scheinleitwertes eines Therapieschwingers; Frequenz in kHz.

trischen Strahlungswiderstand für das bestrahlte Medium und C_0 die Kapazität des festgebremst gedachten Wandlers. Es zeigt sich aber, daß der Teil δ nicht mit dem geometrisch an das Medium angrenzenden der strahlenden Fläche identisch ist, denn die beiden Schwingerpartien sind durch Querkräfte miteinander gekoppelt. Die jeweils tatsächlich strahlende Fläche müßte experimentell bestimmt werden. Eine Berechnung über das vereinfachte Ersatzbild Abb. 3 ist nicht möglich. Dagegen ist

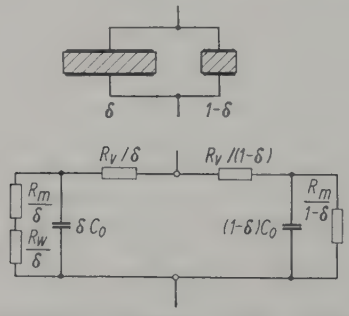


Abb. 3. Elektrisches Ersatzbild eines piezoelektrischen Wandlers bei unzureichender Kopplung an das Medium.

der an den Klemmen des Wandlers auftretende Gesamtwiderstand der Messung zugänglich. Eine entsprechende Kurve des Resonanz-Gesamtleitwertes ist in Abb. 4 als Funktion der kreisabschnittsweise vorgenommenen Benetzung des Therapiekopfes wiedergegeben. Aus seiner Änderung läßt sich eine Überwachung der an den Patientenkörper angrenzenden Fläche des Schallkopfes durchführen. Da es technisch nicht möglich ist, über den ganzen Benetzungsbereich eine konstante Leistung abzustrahlen, kann man der mangelhaften Flächenberührung nur dadurch begegnen, daß man beim Überschreiten einer bestimmten Schwingerleitwertänderung den Therapeuten darauf aufmerksam macht, für hinreichende Flächenberührung zu sor-

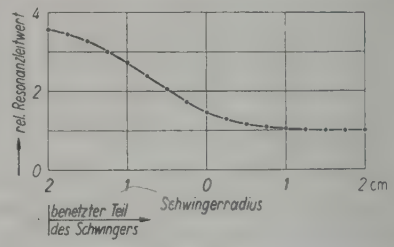


Abb. 4. Resonanzleitwert eines Therapieschwingers als Funktion der kreisabschnittsweise von links nach rechts fortschreitenden Benetzung.

gen. Eine zweckmäßige Lösung besteht darin, daß man bei konstant gehaltener Klemmenspannung die Stromänderung durch den Schwinger für die Steuerung einer Relaisbrücke ausnützt (Abb. 5), die ein akustisches Signal auslöst. Eine solche Steuerung läßt sich speziell beim LSH in technisch einfacher Weise verwirklichen, da die Anpassungsverhältnisse hier besonders günstig sind, wie aus den oben angegebenen Kenndaten hervorgeht. Gleichzeitig wird die Spannung am Wandler auf einen kleinen Wert zurückgeschaltet, damit er nicht überlastet werden kann, solange ein zu großer Teil der strahlenden Fläche an Luft grenzt. Ferner wird die Therapieuhr angehalten, so daß nur die Zeit gemessen wird, in der die vorgewählte US-Leistung in den Körper gelangt. Die kleine Restspannung dient nur noch zur Überwachung des Gesamtwiderstandes und gibt die Möglichkeit, bei hinreichender Flächenberührung von Schwinger und Patientenkörper die Betriebsspannung am Wandler so hoch zu schalten, daß wieder die gewünschte Leistung abgestrahlt wird.

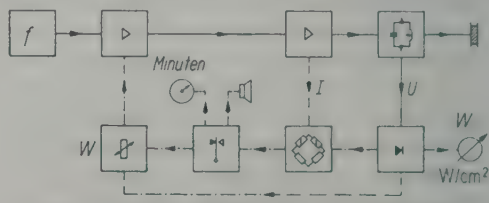


Abb. 5. Prinzipschaltbild eines Therapiegerätes.

Die abgestrahlte Leistung kann bei verschiedener Flächenbenetzung mit einem Strahlungsdruckmesser bestimmt werden. Bei kreisabschnittsweiser Benetzung der strahlenden Fläche ergibt sich Abb. 6, wenn man die erregende Klemmenspannung am Wandler konstant hält. Deckt man also einen Kreisabschnitt ab, der etwa bis zum äußeren Durchmesser der effektiv strahlenden Fläche reicht, so ist die Leistungsabnahme analog der Leitwertänderung

(Abb. 4) verschwindend. Läßt man eine Leitwertänderung von etwa 10% zu, bevor im Therapiegerät die erregende Spannung zurückgeschaltet wird, so resultiert eine Leistungsänderung von ca. 2%. Die abgestrahlte Leistung ist also im zugelassenen Benetzungsbereich praktisch konstant und kann durch Messung der Klemmenspannung bestimmt werden.

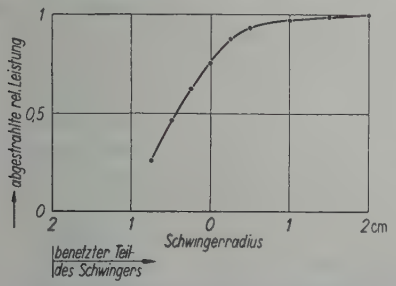


Abb. 6. Abgestrahlte Leistung eines Therapieschwingers als Funktion der Benetzung bei konstanter Klemmenspannung.

Schrifttum

[1] Pätzold, J., Güttner, W. und Bastir, R., Strahlentherapie 86 [1952], 298 ··· 305.

Diskussion

Pahud, J. P.: Comment a été étudié le champ acoustique dont vous avez montré la répartition?

Vortragender: A l'aide d'une sonde mécanique, qui a été décrite récemment par Koppelmann (Acustica 2 [1952], 92).

Oberst, H.: Das zur Messung der Amplitudenverteilung vor dem Wandler benutzte Sondenmikrophon nach Koppelmann ist am schallempfindlichen Ende schallweich und kann deshalb ähnlich wie eine Gasblase in einer Flüssigkeit einen relativ großen Wirkungsquerschnitt haben. Haben Sie bei der Meßfrequenz eine Rückwirkung des Mikrophons auf den Sender bemerkt?

Vortragender: Die Messung der Amplitudenverteilung mit einer schallharten Sonde (sensibilisierte Thermonadel) lieferte das gleiche Ergebnis, also scheint die Rückwirkung gering zu sein.

Richardson, E. G.: Kann man in der Ultraschall-Therapie mit zwei Wandlern arbeiten, um die Energie an einem Ort zu konzentrieren?

Vortragender: Grundsätzlich ist das möglich, aber es ist nicht einfach, eine der geometrischen Optik entsprechende Konzentration zu erhalten, weil die Wandler mit gleicher Phase schwingen müßten. Will man in größerer Tiefe einen Fokus erzeugen, dann muß man die Schallschwächung in Kauf nehmen, die bei den therapeutisch benutzten Frequenzen um 1 MHz sehr groß ist (Halbwertsschicht im Muskelgewebe ca. 3 cm).

THE DESIGN OF LOW FREQUENCY ULTRASONIC TRANSDUCER HOUSINGS

by P. Hatfield

British Nylon Spinners Limited, Pontypool, England

Summary. Ultrasonic techniques have been developed for detecting internal voids and measuring the thickness of rubber products. Low frequency ultrasonic waves from 50 ··· 500 kc/s have been used because of the relatively high absorption of ultrasonic waves in rubber. This paper gives practical details of transducers and their associated water-tight housings, including suitable rubbers for acoustically matching the transducer into a liquid medium, and a simple electrical network for electrically matching the transducer output into an amplifier. It also describes a transmitting and receiving crystal mounted in a common housing for use when only one surface is available.

Sommaire. Procédés de détection des cavités internes des pièces en caoutchouc et mesure de l'épaisseur de ces pièces par les ultrasons (50 à 500 kHz, à cause de l'absorption relativement grande dans le caoutchouc). Détails de réalisations des transducteurs, de leur enveloppe étanche à l'eau et du circuit électrique reliant le transducteur à un amplificateur. Emploi d'un cristal émetteur et récepteur unique si une seule face de la pièce étudiée est accessible.

Zusammenfassung. Ultraschall-Flüssigkeitswandler im Frequenzgebiet von 50 ··· 500 kHz zur Verwendung bei der Prüfung und Dickenmessung von Gummierzeugnissen werden beschrieben, insbesondere Einzelheiten über wasserdichte Gehäuse, darin angebrachte passende Gummiteile zur akustischen Anpassung sowie über die elektrische Anpassung. Weiterhin wird ein als Sender und Empfänger arbeitender Kristallwandler beschrieben, der verwendet wird, falls nur eine Oberfläche zugänglich ist.

1. Introduction

In the development of ultrasonic methods for detecting internal voids in rubber products, particularly tyres, great use has been made of 50 kc/s frequency steel/quartz crystals for transmitting and receiving ultrasonic waves. These crystals mounted in a suitable housing are immersed completely in water for very long periods, and it is essential that the crystal housing prevents ingress of water. After using a variety of crystal housings, one was designed on the principle that if it did leak, it could easily be dismantled, dried out, and put back into service. In practice this type rarely did leak.

The main details of the crystal housing are shown in Fig. 1 and it can be used for circular or rectangular

X-cut crystals. It consists of a brass cylinder and tube as shown. The crystal which is made up of steel/quartz slabs is approximately 45 mm thick and is usually 18 mm in diameter, or 20 × 12 mm² in area. One end of the crystal is stuck on to a rubber bung 10 mm thick, using Dunlop RD/S 203 adhesive. The rubber surface on which the steel electrode of the crystal is to be stuck is cyclised with concentrated sulphuric acid. The adhesive is then applied and the two surfaces are lightly clamped and placed in an oven at about 40 °C. The resultant bond strength of the adhesive is well over 40 kg/cm², and there is little chance of the rubber parting from the crystal. The normal rubber adhesives for cold sticking were unsuitable.

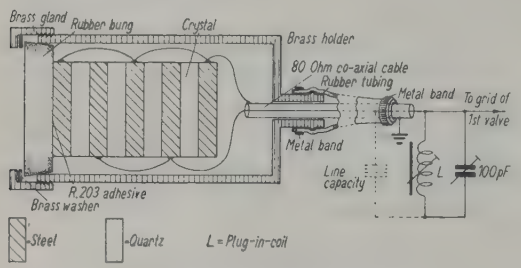


Fig. 1. Low frequency crystal housing.

The co-axial cable is brought right through the narrow neck of the brass housing and is soldered onto the crystal electrodes. The rubber bung and crystal are then forced into the brass holder. Usually a thin coating of polystyrene solution is painted onto the crystal to prevent the possibility of the steel electrodes rusting. The crystal is connected by 3 metres of 80 ohm co-axial cable to its electrical receiver or transmitter. Usually 1 metre of the cable is under water with the crystal housing. To prevent ingress of moisture at the narrow neck of the housing or through the co-axial cable covering which is not always waterproof, a piece of rubber tubing is placed over the cable, and slides over the outside neck of the housing. A slight protrusion on the neck forces the rubber into tension, and the overlap of rubber tubing is held in compression by a metal clip or twisted wire. Because the rubber tends to crack when held in tension, soft plasticised P.V.C. tubing has been used.

The rubber bung on the front of the crystal can be held in position by screwing a brass gland over the rubber, but a brass washer should be placed between the rubber and the gland to prevent distortion of the rubber.

The choice of rubber for most applications is not critical, but its acoustic impedance should be similar to that of water. Natural rubber bungs are fairly soft and possess only a small amount of filler. This rubber has an acoustic impedance approximately equal to that of water, and avoids the necessity of moulding a special rubber compounded solely for its acoustic impedance. It is interesting to note that both natural and synthetic rubbers can be suitably compounded to give a rubber in which the velocity of the waves varies from 1.0 to 2.0 × 10^5 cm/s [1]. Adding carbon black to rubber increases the velocity, but adding iron oxide and other oxide fillers decreases it.

The electrical losses due to the capacity of the cable are considerable and increase with increasing frequency. It is not possible because of space, to match the high electrical impedance of the crystal to the 80 ohm cable in the crystal housing. However, the losses caused by the line capacity can be eliminated by adding a variable inductance and capacity in parallel with the other end of the 80 ohm co-axial cable as shown diagrammatically in Fig. 1. The variable inductance and capacity are mounted in a screened box, and coils of different inductance can be plugged into the box to cover a wide frequency range. It has been found that the 50 kc/s crystals will operate satisfactorily at their odd harmonics up to 1 Mc/s frequency if the line capacity is eliminated.

The semi-angle of the cone of radiation from these crystals at their fundamental frequency (50 kc/s) is approximately 60°. For the practical application of detecting flaws in tyres, where a large area has to be scanned quickly, one transmitter crystal can be used in conjunction with six identical receiving crystals placed side by side [2].

Soft rubber pads on the crystal faces are useful when it is not possible to wet the object under test, e.g. "green" tiles in the ceramic industry. Receiving and transmitting crystals are placed on opposite sides of the object under test and a uniform light pressure is applied. If the tile is free of laminar air voids then a large amplitude signal can be detected on passing ultrasonic waves through the object. If the tile possesses a laminer flaw of area greater than the crystal area, then no signal is picked up. The air gap between the soft rubber and the object is negligible, in fact the soft rubber acts practically as a liquid contact.

Housings for higher frequencies have been built. The rubber face pad has been dispensed with to avoid acoustic losses, but has been replaced by a rubber ring.

2. Common transmitter and receiver crystal housing

A further application of low frequency ultrasonic waves is the measurement of rubber thickness by a

phase lag method when only one surface is available [3]. Here, an identical transmitting and a receiving crystal are mounted side by side, and the receiving crystal must not pick up direct electrical or acoustical signals from the transmitting crystal.

The area of the crystal faces was 1.2×2.0 cm^2, but could be smaller. The two crystals were mounted in a rectangular brass crystal holder approximately $2.7 \times 3.0 \times 6$ cm^3. The crystals were clamped at their mid-point, placed side by side, and electrically screened from each other and from outside electrical radiation. Although the crystals were placed as close together as possible, it was essential that the sides of the crystals did not touch the brass crystal housing; otherwise vibrations would have been picked up by the receiving crystal direct from the transmitting crystal. A co-axial cable was connected to each crystal through a small brass tube as previously described. The top and side of the brass crystal housing were removable to facilitate the mounting of the crystals. The face of each crystal electrode projected slightly from the crystal housing, and the crystals were stuck onto a rubber pad 3 mm thick to prevent any water getting into them. In this application the crystals were not normally immersed in water, but the surface of the rubber under test was thoroughly wetted.

Acknowledgments

Thanks are expressed to Mr. C. CROPPER and Mr. I. FARREN of the General Electric Company, Research Laboratories, Wembley, and Mr. E. F. POWELL, Research Manager, and Mr. R. G. PATTON, Dunlop Rubber Company, for co-operation in this work.

References

[1] HATFIELD, P., Brit. J. appl. Phys. 1 [1950], 252.
[2] HATFIELD, P., Automobile Engineer 41 [1951], 385.
[3] HATFIELD, P., Brit. J. appl. Phys. 3 [1952], 326.

Discussion

FARREN, L. I.: A particular difficulty which is often encountered when transmitting ultrasonic waves through a rubber sheet into water is the effective wetting of the rubber surface. Can the author suggest the surest way of wetting a rubber surface.

LECTURER: We have found that natural rubber moulded free of wax ingredients gives a rubber which wets easily without the help of wetting agents in the water. If mould lubricants are used when moulding the rubber, particularly non-wetting types like silicones, then wetting of rubber is difficult. Cast latex films wet very easily, but we haven't used them extensively because we can get the natural rubber to wet easily.

FEDERICI, M.: Prof. GIACOMINI has shown that there is of course a very effective method of coupling a water-tight housing to water with absolute security and that is by using $\lambda/2$ plate between the transducer and water.

LECTURER: Yes, but the crystal holder described is one type which was found to be water-tight, easily dismantled and easy to make. The holders have been used continuously in an industrial process rather than in a laboratory.

FEDERICI, M.: The measurement of difference of phase, as shown, on a cathode ray tube can be improved by including a delay network on one pair of plates so that the phase difference on the tube is brought to zero; a much more exact value of the existing phase difference can then be read by calibration of the delay network.

LECTURER: Later we were proposing to use a phase shift meter as described by FARREN, L. I., Wireless Engn. 23 [1946], 330. Also Drs. LAWLEY and RICHARDSON, at an earlier meeting in this Congress, described another type of phase shift meter.

LACEY, S. A.: Is it possible to support the transducer at some point along the length to reduce shear stress on the rubber end.

LECTURER: Yes, but this involves practical difficulties in making crystal holder waterproof. No difficulty has been experienced in sticking one end of crystal to rubber pad.

LACEY, S. A.: a) Is the coil used for matching or for some other novel reason? Dr. MASON's book "Electro-mechanical transducers" describes the use of such a coil. b) Has the author used the method of measuring echo repetition rate or frequency of reverberation (for thickness measurement)? The loss in rubber may be too high for this method. c) What is the minimum thickness that can be measured by the phase method.

LECTURER: a) No, we merely mentioned that GEC Research Labs had suggested this method to us in 1947. b) Pulse method was tried but using commercial equipment, the losses in rubber were found to be too high. c) In this application about 4 mm (accuracy $\pm$ 1 mm at 10 mm).

FLORISSON, CH.: Est-ce-que ces dispositifs à cristaux et caoutchouc collés permettent l'émission de puissance assez élevée sans se décoller? Par exemple 2 ou 3 W/cm^2?

LECTURER: These crystals and holders have only been used for low-powered applications less than 0.5 W/cm^2 and if they were used for high powers, the internal heat in the crystal might affect the thermoplastic adhesive between the crystal and the steel electrode.

THE MEASUREMENT OF PHASE AND INTENSITY IN SONIC AND ULTRASONIC FIELDS

by L. E. Lawley and E. G. Richardson
King's College, Newcastle upon Tyne, England

Summary. A method of recording the phase of a received signal in a system of plane progressive waves in relation to another signal of fixed phase is described, with the primary object of measuring changes in the velocity of sound, hence of the composition, of a mixture of gases. Recent work in ultrasonics is reported in which hot-wires and thermistors have been used for intensity measurements in stationary waves or to measure streaming velocities near a quartz generator, to show the usefulness of these devices.

Sommaire. On enregistre la différence de phase entre un signal d'un système d'ondes planes progressives et un autre signal de phase constante en vue de mesurer les variations de la vitesse du son, donc de la composition d'un mélange de gaz. Pour les ultrasons, on a mesuré l'intensité d'ondes stationnaires ou les vitesses de courant près d'un générateur à quartz au moyen de fils chauds et des thermistors.

Zusammenfassung. Mit Hilfe einer Phasenmeßmethode für ebene Wellen werden Schallgeschwindigkeitsänderungen in Gasgemischen verschiedener Zusammensetzung bestimmt. Weiter wird über einige neuere Arbeiten zur Messung der Intensität in stationären Schallfeldern oder der Strömungsgeschwindigkeit vor einem Quarz berichtet, wobei Hitzdrähte und Thermistoren verwendet werden.

1. Measurement of phase

An apparatus has recently been developed in which the proportions of a mixture of two gases can be determined by an acoustic method. A sample of the gas mixture is drawn into a tube about 30 cm long in which there is a telephone earpiece emitting a 3.5 kc/s note at one end and a crystal microphone at the other. If now the gas in the tube is changed from one having a velocity of sound V_1 to one having a velocity V_2 the phase of the voltage at the microphone shifts by $2\pi df\,(1/V_1 - 1/V_2)$, where f is the frequency and d the distance between source and microphone.

A method was required in which the relative phase shift of the voltage at the microphone could be displayed in a manner which could easily be interpreted. The method employed was to amplify the microphone output, square it and then differentiate it. In another amplifier the same operation was done to the sinusoidal voltage applied to the source. The two differentiated outputs were then mixed and the negative pips cut off, so that a series of positive pips appeared, the time interval between which was related to the relative phase angle between source and receiver. This signal was then applied to the Y plates of a cathode ray oscillograph, the time base of which was running continuously at the frequency of the source and was synchronised to start at the same time as the negative differentiated pip from the source. The distance d was now adjusted until the phase shift for the particular gases concerned was just less than 2π. The display was then as in Fig. 1a when the gas of lower velocity filled the tube. When the gas having a higher velocity was introduced, one pip moved to the left until it reached the position of Fig. 1b when the tube was full of the latter gas.

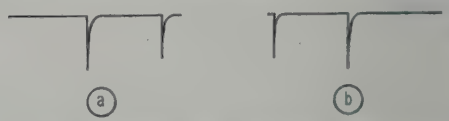


Fig. 1. Pulse technique for measuring phase shifts and velocity of sound.

The centre reference pip was used in this case to mark a particular mixture which was regarded as dangerous. The position of this reference marker may be moved along the trace by applying bias to the source voltage amplifier and causing asymmetrical squaring. This arrangement lends itself easily to the application of an automatic warning system which operates when the movable pip produced by the microphone voltage lies to one side of the reference marker.

Another useful method of displaying the relative phase is to produce a circular time base from the source output voltage and to apply the two pips as brightening pulses. Alternatively the oscillograph display could be discarded and a meter output obtained from the current through a valve which was switched on only during the time interval between the two pips.

The advantage of this method of phase measurement is that since the signal is amplified to a high level before squaring, changes of amplitude due to unwanted stationary waves in the tube or variation of circuit constants do not produce any appreciable effect.

The apparatus is being developed for the use of higher frequencies and consequent reduction in size.

2. Measurement of intensity

For the measurement of intensity, particularly in the ultrasonic range, in a fluid, there is a paucity of apparatus of electro-acoustic type. One of the few pieces of apparatus which is available for this purpose is the hot-wire which, when exposed at a point in an acoustic field, suffers a drop of temperature (therefore, of electric resistance) which can be related by a suitable calibration to the sonic amplitude.

The hot-wire has been applied in our laboratories in a number of ways for the delineation of sonic and ultrasonic fields, some of which will be briefly described.

In a stationary wave system, set up, for example, between a quartz generator and a solid reflector in a gas, it is possible to traverse a hot-wire (heated to about 50 °C above its surroundings) from end to end of the standing wave system, to record the steady drop of resistance experienced by the wire and so to derive the ultrasonic amplitude at, among other places, the nodes and antinodes (Fig. 2). (At low frequencies there is also an alternating fluctuation of resistance at twice the frequency of the sound, but at high frequencies this direct effect is negligible.)

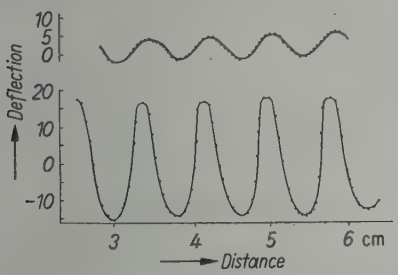


Fig. 2. Stationary wave pattern between a quartz generator and a solid reflector.

From the "standing-wave-ratio" of these amplitudes either the absorption in the fluid or the impedance of the terminals of the column may be calculated; the position of these nodes gives also the ultrasonic wavelength hence the velocity of the waves, all without moving the reflector or source.

3. Measurement of velocity of acoustic streams

Besides these purely acoustic applications, the hot-wire, as is well-known, may be used to measure straight-forward flow. One problem to which we have recently applied it is a borderline case between ultrasonics and hydro-dynamics, i.e. to the acoustic streaming or quartz wind which is set up when the generator works at high power into the fluid. Of course, in this case, there are both alternating (ultrasonic) motions and steady streamings, but the effect on the hot-wire of the latter can be made so large compared to the former that the instrument effectively records only the streaming. In this way we have been able to plot out the vortices set up in the neighbourhood of the source, operated at large amplitudes, and to show how the streaming velocity varies with the frequency for different beam widths (Fig. 3). This has been done for equal power applied to the quartz in several liquids in the hope of relating the results to the "ultrasonic viscosity" which characterises the differing absorption coefficients in liquids.

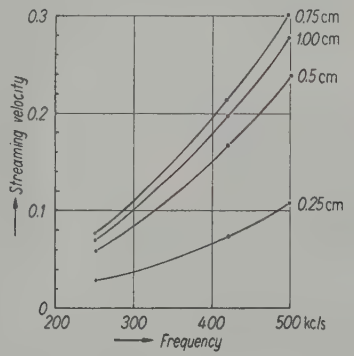


Fig. 3. Streaming velocity as a function of frequency for different beam widths.

It may be pointed out in conclusion that the thermistor (heated semi-conductor element) possesses advantages over the more familiar heated platinum and nickel wire in respect that the element occupies less space — it may be a sphere of about 0.5 mm diameter — and that the sensitivity to ultrasonic amplitude or to steady flow is greater, because the (negative) temperature coefficient of resistance is very high. We have successfully so used thermistors in our experiments.

Discussion

FARREN, L. I.: What accuracy of measurement of phase angle is achieved? Have the authors considered the use of two phase detectors (push pull type) one giving a d.c. output proportional to cos φ and the other an output proportional to sin φ. These outputs may be applied to a crossed coil phase indicator (as used in the Decca Navigator) and indications greater than 2π may be recorded.

LECTURER: The exact figure for accuracy of phase indication is not known but is probably about 1%. The method suggested would be of use only for indicating relative phase changes and would not give any advantage for the practical application intended.

ÉTUDE EXPÉRIMENTALE D'UN TRANSDUCTEUR ULTRASONORE A INCIDENCE OBLIQUE

par G. MALHERBE

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Sommaire. Dans le contrôle ultrasonore des matériaux (surtout dans le contrôle des cordons de soudure), il y a intérêt à attaquer la pièce avec une incidence oblique. La réfraction à l'entrée entraîne une propagation des différents types d'onde (transversal, longitudinal, superficiel) dans des directions différentes. Étude de la répartition de l'énergie entre les différents types d'onde en fonction de l'angle d'incidence. Description du dispositif expérimental et résultats obtenus avec différents aciers.

Zusammenfassung. Bei der Materialprüfung mit Ultraschall arbeitet man oft bei schrägem Schalleinfall. Die Energieverteilung der verschiedenen bei der Brechung des Strahles auftretenden Wellentypen (transversale, longitudinale und Oberflächenwellen) wird untersucht und die Versuchsanordnung sowie einige Ergebnisse werden beschrieben.

Summary. In the ultrasonic testing of materials (particularly in the testing of soldered joints), it is desirable to irradiate the part at an oblique incidence. Refraction at the entrance produces propagation of different types of waves (transverse, longitudinal, superficial) in different directions. Study of the distribution of energy between the various types of waves in terms of the angle of incidence. Description of the experimental device and results obtained with various kinds of steel.

1. Introduction

Dans certains problèmes de contrôle des matériaux, notamment dans le contrôle des cordons de soudure, il y a intérêt à ce que les ultrasons se propagent obliquement dans la pièce à examiner.

Considérons deux tôles A et B assemblées par un cordon de soudure S; la méthode de contrôle ultrasonore par échos permettra de détecter facilement les défauts si les ultrasons se propagent obliquement comme indiqué sur la Fig. 1.

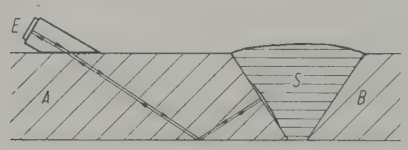


Fig. 1. Trajet des ultrasons dans une tôle soudée.

Pour obtenir cette incidence oblique, la première solution qui se présente à l'esprit est d'intercaler, entre le quartz piézoélectrique et la pièce, un coin de même nature que celle-ci; ainsi le quartz serait directement incliné de l'angle souhaité.

Cette solution n'est toutefois pas acceptable; il est en effet pratiquement impossible d'intercaler pour chaque contrôle un coin d'un matériau identique à la pièce étudiée. Les mesures de célérité effectuées sur différents aciers montrent qu'il ne suffit pas d'avoir un coin en acier quelconque, mais d'un acier rigoureusement identique à celui de la pièce étudiée. Une différence même faible de la célérité des ultrasons dans le coin et dans la pièce entraîne une légère réfraction de l'onde ultrasonore et la séparation dans la pièce des ondes transversales et longitudinales. Il en résulte sur l'écran oscilloscopique une série d'échos parasites qui rendent l'interprétation impossible. Une autre solution consiste à prendre un coin d'un matériau dans lequel la célérité des ultrasons est nettement différente de la célérité dans le matériau à examiner. Dans ce cas, les ondes longitudinales et transversales sont nettement séparées dans la pièce et il y a moyen de choisir l'angle du coin de telle manière qu'il n'y ait que des ondes transversales dans la pièce: il suffit de choisir l'incidence de façon que les lois de réfraction appliquées aux ondes transversales donnent la condition de réflexion totale.

Le but de cette étude est de déterminer l'angle d'incidence pour lequel il n'y a que des ondes transversales dans une pièce d'acier, l'amplitude de ces ondes étant maximum.

2. Dispositif expérimental

Les coins étudiés ont été tirés d'un bloc de Lucite (polymétacrylate).

Une tôle d'acier ABCD de 50 mm d'épaisseur et de 400 mm de côtes a été mastiquée dans une cuve (voir Fig. 2).

Les différents coins étudiés sont appliqués sur la face BC; les ultrasons sortent de la pièce d'acier par la face CD et sont détectés dans la cuve par un récepteur orientable (R) monté sur un chariot. L'axe de déplacement du récepteur est rendu rigoureusement parallèle à la face CD.

L'orientation, la position sur l'axe du déplacement et la distance à la face CD étant connues, il est facile de déterminer la position du point de sortie des ultrasons sur la face CD. Le point d'impact sur la face BC étant également connu, on peut calculer tous les angles de réfraction dès que l'on connaît l'angle d'incidence i et les diverses célérités.

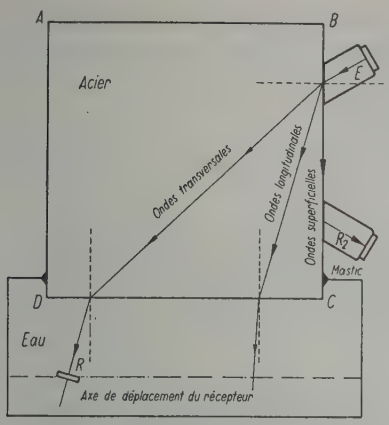


Fig. 2. Dispositif expérimental.

L'appareil utilisé pour ces essais est celui que les ACEÇ fabriquent sous le nom de "Echosonel"; fréquences utilisées: 1 MHz et 2,5 MHz.

La tôle d'acier est recouverte sur toutes ses faces, sauf sur la face CD et au point d'impact des ultrasons, d'un enduit composé de ciment de fer et de graisse et l'intérieur de la cuve est tapissé de caoutchouc rainuré, ce qui assure une certaine atténuation des réflexions parasites.

3. Résultats des mesures

Les mesures de célérité dans les aciers étudiés et dans la Lucite ont été effectuées par la méthode des échos et par résonance. Le Tableau I donne le résultat de ces mesures.

Tableau I: Célérité des ondes longitudinales en m/s.

Matière	Échos	Résonance	Moyenne
Lucite	2550 ± 50	2570 ± 20	2560
Acier I	5830 ± 100	5890 ± 30	5860
Acier II	5670 ± 100	5690 ± 30	5680
Acier III	5430 ± 90	5370 ± 60	5400

	Résistance	Grosseur grain
Acier I	40 kg	gros
Acier II	65 kg	moyen
Acier III	73 kg	fin

Les coins étudiés donnent des incidences de 20°, 25°, 30°, 35°, 40°, 45°, 50°, 52,5°, 55°, 57,5° et 60°.

Pour chaque incidence, une exploration complète est effectuée à l'aide du récepteur R et une mesure des ondes superficielles est également effectuée à l'aide d'un récepteur R_2, identique à l'émetteur E et placé sur la face BC, comme indiqué sur la Fig. 2.

Pour chaque incidence, un graphique analogue à celui de la Fig. 3 est tracé. Dans ce graphique, la longueur des traits est proportionnelle à la hauteur de la déflexion observée sur l'écran de l'appareil.

Ceci permet de déterminer la position de l'axe des faisceaux d'ondes transversales et longitudinales et de déterminer l'angle d'ouverture de chaque faisceau.

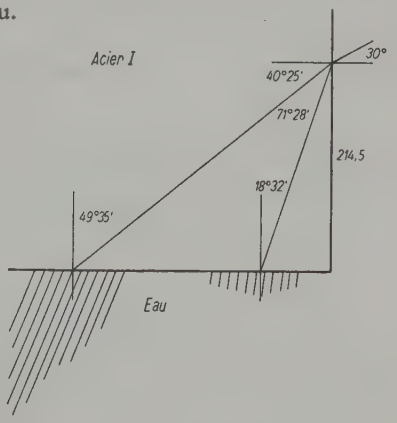


Fig. 3. Graphique type.

Le graphique de la Fig. 4 résume les résultats obtenus pour l'acier I. La hauteur des déflexions caractérise l'amplitude de l'onde.

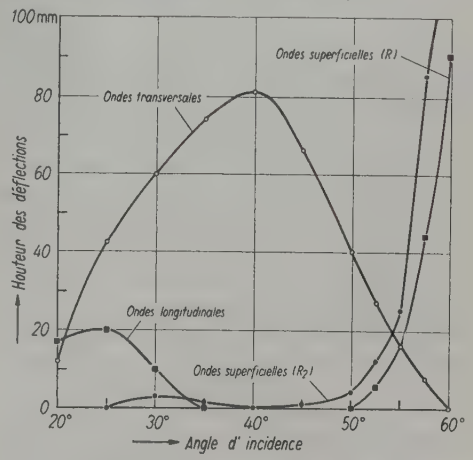


Fig. 4. Transmission des différents types d'onde en fonction de l'incidence.

Ni la nature de l'acier, ni la fréquence ultrasonore utilisée n'ont une influence sensible sur l'aspect de ce graphique. Ceci permet de tirer une conclusion générale:

Un transducteur en Lucite, sur une pièce d'acier, donne:

pour une incidence de 60°: uniquement des ondes superficielles;

pour une incidence de 40°: uniquement des ondes transversales.

Dans ce dernier cas, la propagation dans la pièce d'acier s'effectue sous un angle d'environ 50° par rapport à la normale.

Remarque: La forme du transducteur doit être choisie de manière à éviter les réflexions internes.

USE OF TEMPERATURE- AND TIME-STABILIZED BARIUM TITANATE CERAMICS IN TRANSDUCERS, MECHANICAL WAVE TRANSMISSION SYSTEMS AND FORCE MEASUREMENTS

by W. P. Mason

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Summary. Barium titanate ceramics can be temperature-stabilized by introducing $PbTiO_3$ and $CaTiO_3$. Two stable compositions have high electrical and mechanical "*Q*'s". Ageing occurs because of domain wall motions induced by residual stress. By heating to a temperature under the Curie point, faster ageing occurs. External stress causes ageing and high stresses completely depolarize the ceramic, producing a charge one third that for single domain crystals. A rate theory accounts for the results.

Sommaire. On peut, grâce à des additions de TiO^3 Pb et de TiO^3 Ca, stabiliser les produits céramiques au titanate de baryum par rapport à la température. Deux compositions stables ont des valeurs élevées du "Q" électrique et mécanique. Il se produit un vieillissement dû aux tensions résiduelles. Il est accéléré par chauffage à une température inférieure à celui du point de Curie. Un effort mécanique extérieur produit un vieillissement, et, s'il est important, dépolarise complètement le produit céramique. Une théorie rend compte des résultats des mesures.

Zusammenfassung. Durch Einlagerung von $PbTiO_3$ und $CaTiO_3$ können Bariumtitanate temperaturstabilisiert werden, wobei zwei Zusammensetzungen hohe elektrische und mechanische Güten haben. Durch Restspannungen und auch durch äußere Spannungen treten infolge Verschiebung der Domänenwände Alterungseffekte auf, die durch Erhitzen noch beschleunigt werden. Starke äußere Spannungen depolarisieren das Material völlig.

Ceramics made from barium titanate have been used extensively in electrical condensers and in electromechanical transducers. The properties of interest are dielectric, effective piezoelectric and elastic constants and these are shown by Fig. 1 for commercial barium titanate. These properties change markedly at the Curie and second transition temperatures and are relatively constant between them. Many more uses could be satisfied if these properties were less temperature-sensitive and if the time stability were good. Some of these uses are in force measurements, in electrical and electromechanical filters and in wave propagation systems.

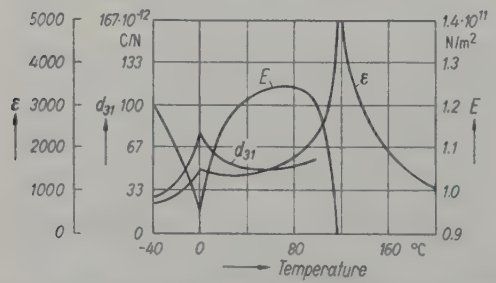


Fig. 1. Dielectric (ε), effective piezoelectric (d_{31}) and elastic (E) constants of commercial barium titanate as a function of temperature.

It is well known that additions to barium titanate ceramics change the positions of the critical temperatures. In particular, $PbTiO_3$ [1] raises the Curie temperature and lowers the second transition temperature while $CaTiO_3$ [2] lowers the second transition temperature without changing the Curie temperature. Neither additant alone gives the characteristics desired for a filter material, namely temperature stability and a high "*Q*", but it has been found that ceramics made from both additants produce stable high *Q* ceramics. Fig. 2 shows the

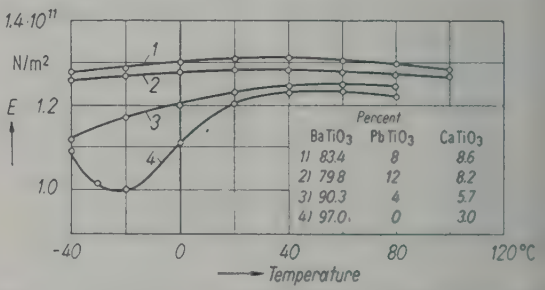


Fig. 2. Improved temperature stability of titanate combinations.

value of Young's modulus E for a number of combinations measured from -40 °C to $+90$ °C. There are two combinations 80% $BaTiO_3$, 12% $PbTiO_3$, 8% $CaTiO_3$ and 84% $BaTiO_3$, 8% $PbTiO_3$, 8% $CaTiO_3$ whose moduli peak at about 25 °C and whose variation is about 0.2% (0.1% in frequency) over the normal indoor temperature range. The piezoelectric and relative dielectric constants are rather stable and have aged values $d_{31} = 2.2 \times 10^{-11}$, $d_{31} = 1.75 \times 10^{-11}$ coulomb/newton and $\varepsilon_{33}^T = 500$, $\varepsilon_{33}^T = 412$ respectively for the 84-8-8 and 80-12-8 combinations. The d_{33} piezoelectric constants are 2.6 times d_{31}. The *Q*'s of the dielectric and elastic

constants are 400 and 1600. Hence, these ceramics have good enough mechanical and electrical properties for use in filters and wave transmission systems.

All of these ceramics show a pronounced ageing effect during which the dielectric constant decreases from 5 to 20 per cent, the elastic constant increases from 1 to 4 per cent, the effective piezoelectric constant decreases from 5 to 25 per cent and the Q increases by factors as high as three. The lower figures quoted are for ceramics containing 4% $PbTiO_3$, which age less than other compositions. Fig. 3 shows the frequency variations Δf of the two

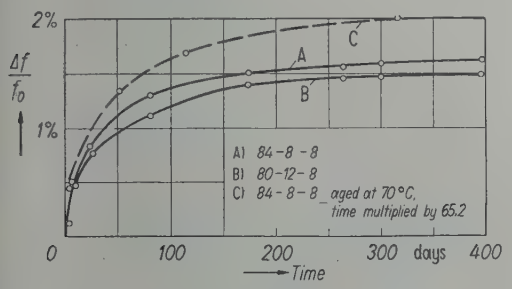


Fig. 3. Frequency variation (ageing) of two temperature stable ceramics at 25 °C (curves A and B); for comparison curve C at 70 °C (extended time scale).

temperature-stable ceramics as a function of time for 25 °C. The frequency follows a formula

$$f - f_0 = K \frac{A}{1 + At} \qquad (1)$$

where K and A are constants and t is the time. Ceramics were heated and repoled, and went through a similar ageing cycle. This shows that the ageing is caused by conditions set up when the ceramic becomes ferroelectric. As shown by Fig. 4a, each grain has at least six domains with the adjacent polarizations at right angles. Since there is an extension parallel and a contraction perpendicular to the direction of polarization, the domains will not fit together unless they are strained. In addition, X-ray and temperature expansion measurements show that a system of radial tensions exist below the Curie temperature as shown by Fig. 4d. If the adjacent

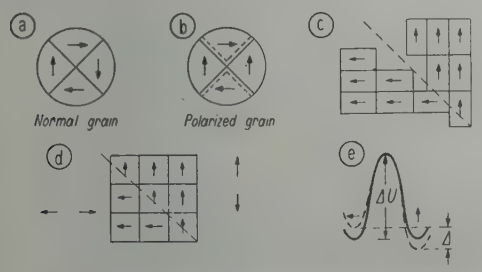


Fig. 4. Concerning the polarization mechanism.

domains are of exactly the same size, the stresses are balanced and the domain wall will be stationary. Actually the grains are irregular in size and have residual stresses so that adjacent stresses are not equal and domain wall motions will occur by unit cells changing direction by 90° until the stresses on the two sides of the wall balance. This process is probably the cause of dielectric and elastic ageing of unpolarized ceramics. On poling, one of the six domains reverses direction, leaving a small region to minimize electrostatic energy, and also domain walls move. This builds up a stress pattern in the smaller sized domains, and when the poling voltage is removed, the domain walls move back with a reduction in the effective piezoelectric constant. When a unit cell changes direction, it crosses an energy barrier of the type shown by Fig. 4e. Hence, the ageing process can be speeded up by heating a ceramic to a higher temperature under the Curie point. The dashed line of Fig. 3 shows the ageing at 70 °C when the time is multiplied by 65.2. This factor corresponds to an activation energy of 19 kilocalories (= 80 kJ) per mole. When ceramics were aged for three weeks at 70 °C and one week at 25 °C, the remaining frequency variation was less than 0.15%. No further ageing is observable in dielectric or piezoelectric constants or in the Q.

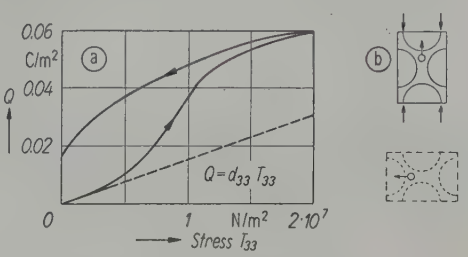


Fig. 5. Polarization by external stress. Dotted: relation for low stresses. Drawn in full: higher stresses and hysteresis.

External stresses applied in the direction of polarization cause the charge developed to follow initially the piezoelectric relation *) $Q = d_{33} T_{33}$, but for high stresses the charge is larger and becomes asymptotic to 5.7×10^{-6} coulombs/cm², one-third the saturation charge for a single domain crystal (Fig. 5). This is the result of depolarizing all the Z unit cells by changing the shape as shown by Fig. 5b. When one takes the stress off (return curve of Fig. 5), the ceramic becomes partially depolarized. A constant stress of 3.5×10^7 newtons/meter² causes a decrease in the piezoelectric constant as shown by Fig. 6. This is caused by domain wall motions which cut down the size of the polarized domains.

*) The symbol Q is used both for quality and for charge!

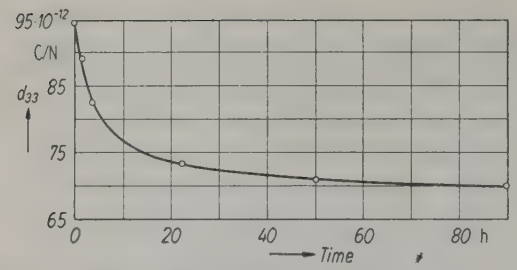


Fig. 6. Decrease of piezoelectric constant at 25 °C by a constant stress of 3.5×10^7 N/m².

A rate process theory has been developed which agrees well with the measured data. The frequency equation (1) results if the stress difference in adjacent domains is equal to $T_{33o}\,(X/L)^2$ where X/L is the ratio of the distance of the domain wall from its equilibrium position to the domain size and T_{33o} is the stress necessary to squeeze the tetragonal-shaped unit cell to a cubic form. The rates of ageing due to impressed and residual stresses are consistent with the activation energy of 19 kilocalories (=80 kJ) per mole measured by the relative ageing rates.

References

[1] Mason, W. P., Piezoelectric crystals and their application to ultrasonics. Van Nostrand, Inc., New York 1949, chapter 12, p. 308.

[2] Berlincourt, D. A. and Kulcsar, F., J. acoust. Soc. Amer. **24** [1952], 709.

Discussion

Florisson, C.: If I understand correctly, for applications it is advantageous to use these ceramics containing high doses of Ca and Pb titanates, to make them more stable with time.

Lecturer: Yes; ceramics whose critical temperatures are outside of the operating temperature range are less subject to thermal hysteresis effects. Furthermore, the internal losses are less.

Federici, M.: I should like to have it quite clear that ageing is best done at a higher temperature (70 °C) than at a lower one.

Lecturer: Yes; the higher temperature causes an increase in domain wall velocities by a factor of 65 to 1 and a faster ageing.

Brown, R. E. C.: When used as a delay element, what is the loss between input and output circuits?

Lecturer: The losses are from 20 tot 30 dB, depending on the mode and manner of driving.

Skudrzyk, E.: What is the shortest pulse that could be stored in a barium titanate memory cell as used in electronic calculators?

Lecturer: Memory cells are usually made from single crystals rather than the ceramics discussed in this paper. Their storage time is 1 μs or less as discussed by J. R. Anderson, Electr. Engng., **71** [1952], 916.

Lamb, J.: Do you have any information concerning the possible effect of absorbed water vapour in the ceramic on the mechanical and dielectric properties.

Lecturer: With densities of 93% or higher of the crystal density, the porosity is very low and the effect of water vapour on the dielectric constant is small.

Nödtvedt, H.: How does the frequency-stability with temperature of permendur, where coupling coefficients of 40% may be reached, compare with barium titanate with coupling coefficients of 20%?

Lecturer: High coupling permendur does not have as good temperature coefficients as the stabilized barium titanate and furthermore is limited in frequency by microeddy current effects.

Farren, L. I.: In view of the great possibilities of improved design of electrical filters due to the high electromechanical coefficient of barium titanate and its variations (as compared with quartz), is the stability of the material ever likely to be made good enough for this purpose?

Lecturer: The titanates in their present form are not nearly as stable as quartz. They are stable enough to be considered for certain types of filters and on account of their low cost may be used in certain filter structures.

AN ACOUSTIC WAVE GUIDE LENS FOR USE IN LIQUIDS

by H. Nödtvedt

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Summary. A short treatment of sound propagation between parallel, completely resilient plates is given. The phase conditions inherent in a medium of this kind make it possible to construct acoustic lenses in liquids. Measurements of the frequency characteristics of a set of plates show a high-pass filter response, in agreement with theory. The data for a lens to be used in water at 30 kc/s are given, and its directive properties presented.

Sommaire. On étudie brièvement la propagation du son entre des plaques parallèles, parfaitement élastiques. Les conditions de phases régnant dans un milieu de ce genre rendent possible la réalisation de lentilles acoustiques dans les liquides. D'après les résultats de la mesure des caractéristiques en fréquence d'une série de plaques, celles-ci se comportent comme un filtre passe-haut, conformément à la théorie. On donne les caractéristiques de la lentille à employer dans l'eau pour 30 kHz, et on indique les propriétés directives de cette lentille.

Zusammenfassung. Die Schallausbreitung zwischen völlig schallweichen Platten wird kurz behandelt. Die Phasenverhältnisse in einem solchen Medium gestatten die Konstruktion einer akustischen Linse in Flüssigkeiten. Durch Messungen an einem Satz schallweicher Platten wird der Hochpaß-Charakter in Übereinstimmung mit der Theorie bestätigt. Für eine Wasserschall-Linse bei 30 kHz werden einige Daten und die Richtwirkung angegeben.

1. Introduction

The problem of concentrating sound energy into narrow beams is very similar to the corresponding problem in the electromagnetic short wave field. The mathematical methods used are in many cases identical and, in some cases, the same physical device may even satisfy the requirements in both fields if the wavelengths involved are the same. To achieve narrow beams, lenses of various types have been introduced. KOCH [1], [2], [3], BROWN and JONES [4] and others give descriptions of electromagnetic and acoustic lenses based on path-length differences, a symmetrical arrangement of perfectly reflecting obstacles and, in the electromagnetic case, also on the phase conditions inherent in wave guides consisting of parallel, perfectly conducting sheets.

While the two former principles both have electromagnetic and acoustic versions, the wave guide action does not lend itself to acoustic application as long as only gases are considered as media of propagation. In liquids, however, the determining physical constants are such as to make lenses of this type very practical and easily constructed.

2. Wave propagation between parallel plates

We assume that plane sound waves of a single frequency, with the wave front parallel to the Y-axis, fall upon a number of parallel plates of negligible thickness (Fig. 1). The plates are spaced a distance a apart, and they are assumed to have an infinite extension in the Y-direction. They are, further, supposed to be made of completely resilient material, such that the acoustic impedance of the plates is negligible compared with that of the medium. The wave equation for the propagation between the plates, takes the form:

$$\frac{\partial^2 \Phi}{\partial x^2} + \frac{\partial^2 \Phi}{\partial z^2} = \frac{1}{c^2}\frac{\partial^2 \Phi}{\partial t^2} \tag{2.1}$$

the solution of which is:

$$\Phi = \sum_{m=0}^{\infty} A_m \sin k_{1m} x \, (B_m e^{j k_{2m} z} + C_m e^{-j k_{2m} z}) \, e^{j\omega t} \tag{2.2}$$

where $k_{1m}^2 + k_{2m}^2 = k^2 = (\omega/c)^2$.

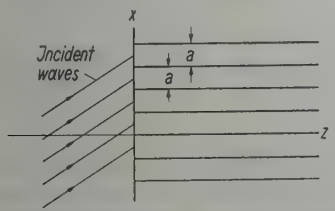


Fig. 1. Waves incident on a set of parallel plates.

If, as is assumed, the plates are completely resilient, the sound pressure, $p = \varrho \partial\Phi/\partial t$, must vanish at the walls, i.e. for $x = 0$ and $x = a$. This defines the eigenvalues as:

$$k_{1m} = m\frac{\pi}{a}, \; (m = 1,2,3, \cdots)$$
$$k_{2m} = \sqrt{\left(\frac{\omega}{c}\right)^2 - \left(\frac{m\pi}{a}\right)^2}. \tag{2.3}$$

In order to allow propagation in the Z-direction, k_{2m} must be real, and consequently for each value of m, the minimum frequency for which transmission will take place, is given by: $\omega_m = mc\pi/a$. As the frequency is increased, more modes will be transmitted corresponding to $m = 1,2,3, \cdots$ The wavelengths corresponding to the different limiting frequencies are found from: $\omega_m/2\pi = f_m = mc/2a = c/\lambda_m$, where $\lambda_m = 2a/m$. The longest wavelength to be transmitted is thus twice the distance between the plates. The zero order mode does not appear in the solution, since it cannot be sustained by the resilient walls.

The phase velocity of the mth order mode is found from:

$$v_z = \frac{\omega}{k_{2m}} = \frac{c}{\sqrt{1 - \left(\frac{f_m}{f}\right)^2}}. \tag{2.4}$$

For every incoming wave of a certain frequency there will be a certain number of modes propagated, corresponding to the limiting frequencies. For modes higher than a particular number m, no energy will be propagated, but the so called evanescent waves will appear. These evanescent waves are made up of exponentially decaying pressure and velocity fields, and they are attenuated very quickly. When designing lenses, the spacing between the plates is chosen such that only one mode is being propagated.

3. Lens design

The phase conditions inherent in the plate medium, as shown by eq. (2.4), make it possible to construct lenses in liquids, where materials with negligible acoustic impedances, compared to the medium, can be found. The conditions for equiphase at the line a-a from a sound source at P in fig. 2, leads to the formula for the inner surface of the lens, if the outer surface is plane:

$$x^2 - 2(1 - n) f_f z - (1 - n^2) z^2 = 0 \tag{3.1}$$

where n = refractive index $= c/v_z$, and f_f = focal length.

Apart from determining the "optical" properties, the refractive index is also included in the expression for the input impedance and consequently determines the minimum length for which the evanescent waves may be neglected. It is also of importance when the reflections at the entrance to the lens are

considered. These reflections are extremely complex. The corresponding problem of reflections of short electro-magnetic waves from metallic plate media has been treated by CARLSON and HEINZ [5], BERZ [6] and WHITEHEAD [7], and the procedure may without alterations be transferred to the acoustic field. The practical conclusions are that the entrance reflections are "optical" and small (less than 10% of energy) as long as the angle between the incident waves and the normal to the plate medium is less than a so-called critical angle. This critical angle depends on the refractive index. For greater angles of incidence, higher order reflections occur, and the efficiency of the lens deteriorates.

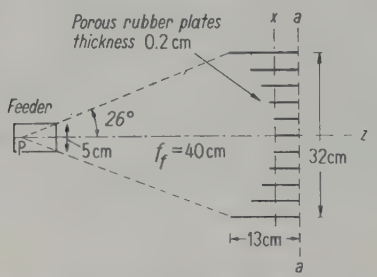


Fig. 2. Lens design at 30 kc/s. Focal length = 40 cm, refractive index = 0.6, plate spacing 3.13 cm.

The data for the cylindrical lens used in the present case are given in Fig. 2. The sound source (feeder) has a dimension in the plane of interest of one wavelength and consequently a half value angle of 25°. This gives good covering of the lens aperture.

4. Measurements

To measure the transmission properties of a plate medium, 10 plates with dimensions 10 cm × 30 cm were cut from 0.2 cm thick sorbo rubber. They were mounted parallel in a frame, and this frame was then inserted between a sound source and a small sound receiver. The distance from the plates to the receiver was kept small to prevent pick-up due to refractions round the plates. The frequency response of source and receiver was measured with and without the plate medium inserted and the transmission characteristics of the plates worked out. Figs 3a and b show the attenuation versus frequency. The theoretical cut-off frequencies marked are seen to agree with measured values. Since the dimensions of the plate medium at the lower frequencies are only two or three times the wavelengths, the signals picked up by the microphone at these frequencies may be due to refraction.

Next, the directive properties of the lens given in Fig. 2 were measured at 30 kc/s. The feeder, which was a magnetostrictive transducer with a vibrating area of 5 cm × 25 cm, was first placed in the lens

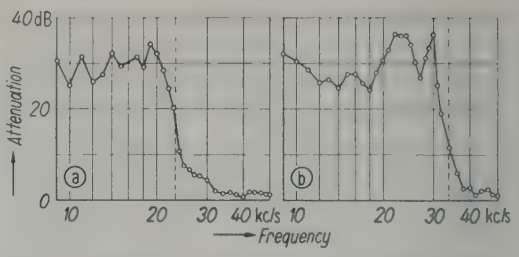


Fig. 3. Transmission characteristics for plate media.
a) Plate spacing 3.2 cm, theoretical cut-off frequency 23.5 kc/s;
b) Plate spacing 2.2 cm, theoretical cut-off frequency 34 kc/s.

axis at the focal point. The polar diagram was then measured as shown in Fig. 4 where also the polar diagram of the feeder alone is plotted. The increase in gain was measured to 6 dB, as compared to a theoretical value of approx. 7.8 dB. It must be assumed that a certain loss of energy occurs along the passage between the plates, and in addition some energy is reflected at the surface of the lens, such that the agreement is reasonably good. It should be possible to reduce the loss due to reflections at the lens surfaces, by inserting impedance matching plates whose characteristic impedances are the geometric mean of the impedance in the medium and the lens impedance. Rubber with the required properties is available, but it is likely that inner frictional losses in these matching plates will make the additional gain negligible, or even reduce it.

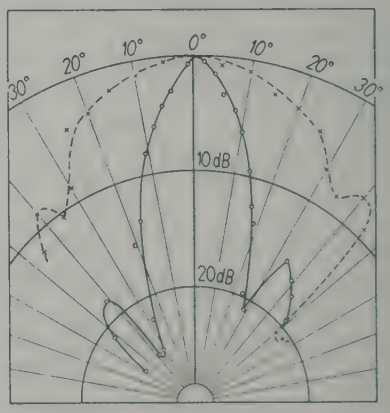


Fig. 4. Measured directivity curve of lens at 30 kc/s. Plate spacing 3.13 cm. Dotted curve shows directivity of feeder alone.

The influence of moving the feeder off the axis was then measured. Keeping the focal length, the feeder was displaced 3 cm on either side of the plane of symmetry of the lens and the polar diagrams were plotted. The results are shown in Fig. 5 and, as will be seen, no major change occurs in the main

lobe. The secondary lobes, however, show a certain increase. The same effect was noted when the feeder was moved along the lens axis.

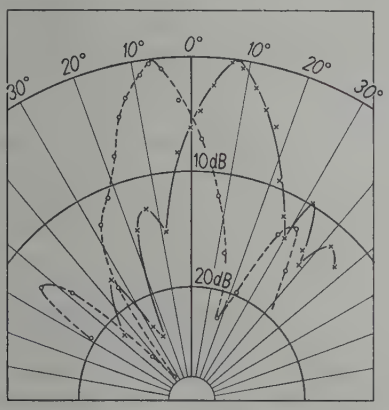


Fig. 5. Directivity curves of lens when feeder is displaced 3 cm to the right (resp. to the left) of the lens axis.

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[2] Kock, W. E., Bell Syst. tech. J. **27** [1948], 58.
[3] Kock, W. E., Proc. Inst. Radio Engrs. **37** [1949], 852.
[4] Brown, J. and Jones, S. S. D., Electronic Engng. **22** [1950], 127.
[5] Carlson, J. F. and Heinz, A. E., J. appl. Math. **4** [1947], 313.
[6] Berz, Feodora, J. Instn. electr. Engrs. **98** [1951], 47.
[7] Whitehead, E. A. N., J. Instn. elect. Engrs. **98** [1951], 133.

Discussion

Brittain, F. H.: At what angles of incidence will total reflection occur?

Lecturer: Total reflection in the optical sense does not occur, but at angles of incidence greater than the critical value, higher order reflections appear which seriously reduce the efficiency of the lens. The critical angle of incidence varies with the refractive index and will for instance increase from 25° to 65° when the refractive index is decreased from 0.7 to 0.3. The entrance reflections will however also increase with decreasing refractive index due to increasing mismatch.

ÉTUDE EXPÉRIMENTALE DU CHAMP DE TRANSDUCTEURS ULTRASONORES FOCALISANTS

par J. Ph. Pahud

Laboratoire de Recherches Physiques des A.C.E.C., Charleroi, Belgique

Sommaire. Étude au moyen d'une sonde thermoélectrique du champ rayonné dans l'eau aux fréquences 0,5, 1, 2, 3 et 4 MHz par des transducteurs munis de pavillons focalisants biconiques.

Zusammenfassung. Anwendung einer thermoelektrischen Sonde zur Untersuchung eines von fokalisierenden doppelten Hohlkegeln erzeugten Ultraschallfeldes in Wasser bei den Frequenzen 0,5, 1, 2, 3 und 4 MHz.

Summary. Use of a thermoelectric probe to study the field radiated in water at 0.5, 1, 2, 3 and 4 Mc/s by ultrasonic transducers with biconical focussing horns.

On sait que si l'on munit une source ultrasonore d'une pièce ayant la forme de deux troncs de cône accolés par leurs sections de petit diamètre (Fig. 1 et 2), on obtient une focalisation du faisceau d'ultrasons autour d'un foyer F: cet effet résulte de la double réflexion des rayons issus de la couronne AB-CD de la source S. Barone [1] a décrit récemment de tels dispositifs et a indiqué la façon de calculer des "pavillons" de ce type. Le calcul est basé sur des considérations d'acoustique géométrique et n'est valable par conséquent que lorsque la longueur d'onde est petite par rapport aux dimensions du pavillon, c'est-à-dire pour des ultrasons de haute fréquence. Nous avons construit deux pavillons, de distance focale théorique 18 mm et 91 mm et nous avons étudié le champ acoustique qu'ils fournissent dans l'eau lorsqu'ils sont montés sur des transducteurs piézoélectriques (quartz X) travaillant aux fréquences 0,5, 1, 2, 3 et 4 Mc/s.

Nous avons utilisé pour cela une sonde thermoélectrique perfectionnée ("Microsonel", brevet A.C.

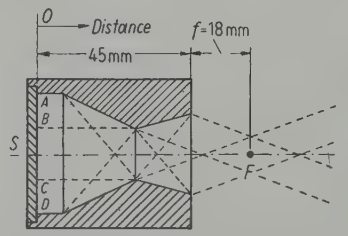


Fig. 1. Transducteur muni d'un pavillon focalisant biconique. Distance focale théorique = 18 mm. S = source; F = foyer.

E.C.) dérivée de la sonde décrite par Pinoir [2]. Elle se compose d'un thermocouple cuivre-constantan dont la soudure "chaude" est enrobée dans une sphérule de matière absorbant les ultrasons (polyester Marco 28), tandis que la souduré "froide" est placée au voisinage immédiat de la soudure chaude et baigne dans le milieu de propagation, contrairement à la technique usuelle des mesures thermoélectriques où la soudure froide est plongée dans

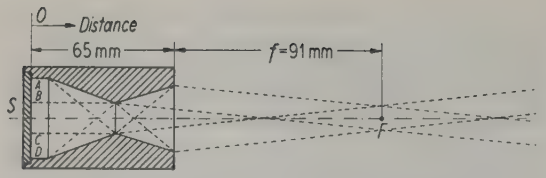


Fig. 2. Transducteur muni d'un pavillon de distance focale théorique 91 mm.
S = source; F = foyer.

de la glace fondante. Cette disposition a l'avantage de rendre la force électromotrice du thermocouple indépendante des fluctuations de température d'origine étrangère et de la faire dépendre uniquement de la différence de température entre le milieu de propagation et la sphérule qui s'échauffe en absorbant les ultrasons. Un microtube en acier inoxydable assure à la fois la rigidité mécanique de la sonde et le blindage électrique du circuit. La sphérule a un diamètre de 1 mm, la distance entre les soudures est de 5 mm, les fils de cuivre et de constantan ont un diamètre de 0,1 mm, celui du microtube est de 0,8 mm. On a ainsi une sonde extrêmement réduite qui d'une part permet de faire des mesures ponctuelles et qui, d'autre part, ne perturbe pas le champ acoustique. La force électromotrice qui prend naissance entre les soudures, de l'ordre de 10^{-5} V, est amplifiée par un amplificateur à courant continu PERKIN-ELMER, ce qui nous a permis d'atteindre une très bonne sensibilité jointe à une bonne stabilité.

La sonde est fixée à un dispositif de chariotage qui permet de la déplacer dans le champ. On a relevé au préalable la loi suivant laquelle la force

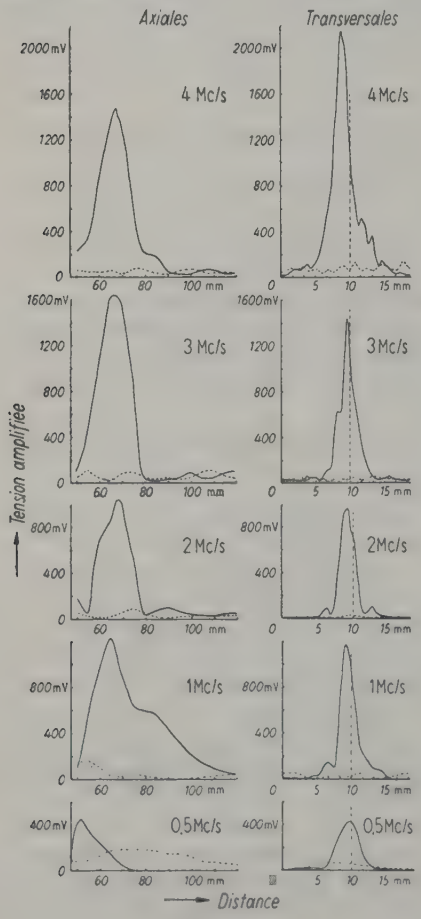


Fig. 3.

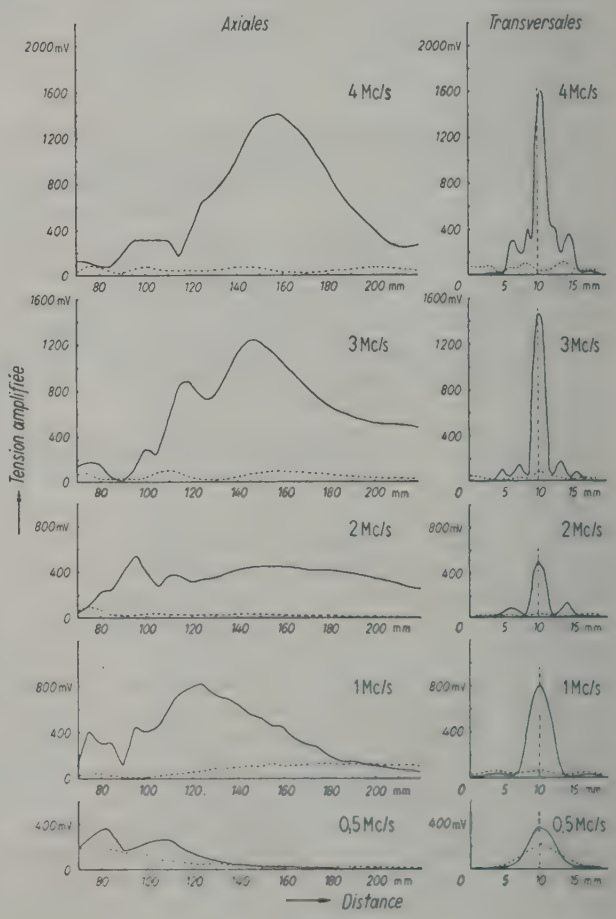


Fig. 4.

Fig. 3 et Fig. 4. Exploration du champ acoustique suivant l'axe du faisceau ultrasonore et suivant une direction perpendiculaire à l'axe, au droit du foyer réel.
——— avec pavillon; ---------- sans pavillon.
Dans les diagrammes d'exploration transversale, l'axe géométrique du faisceau se situe à l'abscisse 10 mm.
Fig. 3 : f = 18 mm; foyer théorique à 63 mm de la source.
Fig. 4 : f = 91 mm; foyer théorique à 156 mm de la source.

électromotrice fournie par le thermocouple dépend de l'intensité ultrasonore, mesurée en valeur relative par le carré de la tension HF appliquée aux faces du piézoquartz. Cette caractéristique a été relevée à chacune des fréquences utilisées, car l'absorption de la sphérule dépend de la fréquence.

L'étude a porté sur deux transducteurs équipés des pavillons mentionnés plus haut. Les transducteurs étaient montés dans la paroi d'une cuve anti-reflet remplie d'eau, les quartz ainsi que les pavillons étant aisément interchangeables. Nous avons procédé à une exploration suivant l'axe du faisceau ultrasonore, ce qui nous a permis de localiser le foyer; puis suivant une direction perpendiculaire à l'axe et passant par le foyer. Après chaque exploration, nous avons démonté le pavillon et relevé, suivant les mêmes axes de déplacement, le champ rayonné par le transducteur privé de la pièce focalisante.

Les Fig. 3 et 4 récapitulent lès résultats obtenus et illustrent les conclusions suivantes:

1. L'effet focalisant des pavillons à double tronc de cône est très net.
2. Pour le pavillon à courte focale, le foyer est bien localisé à l'endroit calculé. Sa position est indépendante de la fréquence, sauf à 0,5 Mc/s où la distance focale est plus courte que prévu.
3. Pour le pavillon à grande focale, le foyer n'est à l'endroit calculé que pour la fréquence de 4 Mc/s. Quand la fréquence diminue, le foyer se rapproche du pavillon en même temps que sa structure se complique de maxima secondaires.

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[2] PINOIR, R., J. Chim. phys. 44 [1947], 254.

Discussion

BRADFIELD, G.: An alternative method which we have found highly effective for focussing ultrasonic energy has been indicated in Fig. 2C of the General Review which illustrates a barium titanate toroïd faced by a concave mirror (air-backed) so that the beam is concentrated through the centre of the toroïd. Very intense beams can be so produced and the troubles mentioned by the author from large side lobes do not seem to arise.

CONFÉRENCIER: There are many other methods to concentrate ultrasonic energy, and the one indicated by Mr. BRADFIELD has the advantage of giving a real focus and not a focal zone such as the method described in the paper.

POIANI, G.: Est-ce que vous n'avez pas eu des perturbations électrostatiques dans vos mesures?

CONFÉRENCIER: Non, car le blindage électrostatique était assuré par l'eau, qui est bonne conductrice, et par le microtube d'acier, qui entourait les fils du thermocouple. Mais nous avons observé des perturbations sérieuses lorsque la soudure froide était mal isolée; ces perturbations sont dues au potentiel de contact entre l'eau et le métal.

LEVI, F. A.: a) Was the device able to detect the fine structure? There was of course an integration on the absorbing drop. b) How was it possible to see that no diffraction was responsible for the irregularities of the longitudinal intensity diagram?

CONFÉRENCIER: a) The device was able to detect standing waves of 0.75 mm interspace, although with a smoother curve due to integration on the drop. With such a "separating power" it should be possible to detect the fine structure resulting from the two beams intersecting obliquely in the focal zone. This fine structure was not observed, though the theoretical interfringe with the 91 mm double cone at 4 Mc/s was 1.7 mm. The fine structure seems to be disturbed by other phenomena which are responsible for the irregularities in the axial repartition of intensity. b) By calculating the diffraction from the smaller section of the double cone. The reason for the irregularities is the diverging of the beam from the crystal. We are here at the limit of validity of geometrical acoustics and the only correct method is to solve the partial differential equations with the boundary conditions on the conical surfaces.

FLORISSON, C.: a) Les cônes sont-ils à double enveloppe remplie d'air? b) Je signale qu'avant PINOIR, le Professeur DOGNON, vers 1932, employa le couple avec sphérule absorbante dans les mesures de champs ultrasonores.

CONFÉRENCIER: a) Non, en laiton plein, qui sous ces incidences donne la réflexion totale. b) La méthode du thermocouple sensibilisé par un absorbant a été utilisée depuis longtemps par maint chercheur, mais seuls PINOIR et nous prenons comme référence la température de l'eau au voisinage immédiat du point de mesure.

SOME INDUSTRIAL APPLICATIONS OF ULTRASONICS

by R. W. SAMSEL, G. E. HENRY and R. O. FEHR
General Electric Co., Schenectady, N.Y., U.S.A.

Summary. Today cleaning of small metal parts is the most important application of high power ultrasonics in industry. Studies have been made to determine the requirements of intensity and exposure time for cleaning operations. An adequate transducer has been built and installed in a factory. A probe, consisting of a thermistor imbedded in a small plastic sphere, has been developed for measuring intensity.

Sommaire. Application d'ultrasons de grande puissance dans l'industrie: nettoyage des pièces. Dans ce cas on a déterminé l'intensité sonore et la durée de traitement nécessaire. On a construit à cet effet un générateur d'ultrasons. Pour mesurer l'intensité, on a mis au point une sonde thermistor noyé dans une petite sphère en matière plastique, qui absorbe l'énergie ultrasonore et la convertit en chaleur: la température est proportionnelle à l'énergie ultrasonore absorbée par la sonde.

Zusammenfassung. Im Gegensatz zu Ultraschallgeräten geringer Leistung finden solche mit größerer Leistung bisher in der Industrie noch wenig Anwendung, ausgenommen zur Reinigung von kleinen Metallteilen. Die hierfür notwendigen Intensitäten und Beschallungszeiten werden untersucht. Zu diesem Zweck ist eine Sonde zur Messung der Schallenergie entwickelt worden, die aus einem in eine kleine Kunststoffkugel eingebetteten Thermistor besteht, dessen Temperatur der in der Kugel absorbierten Energie proportional ist.

1. Introduction

In the field of electrical engineering, a distinction is made between power applications and application for communication or measurement purposes. A similar division may be made in the field of ultrasonics. One area of operation, that of communication, signalling, and gaging, includes such applications as flaw detection, thickness measurement, and measurement to determine some of the characteristics of liquids. These applications have been put to practical use many years ago, and their usage is well-established. Many suggestions for the use of high power ultrasonics have been made such as cleaning of metal parts, cleaning of fabrics, smoke precipitation, and mixing of paints. However, in the class of power applications to date, cleaning of small metal parts is the only one in general use in industry in the U.S.A. Possibilities for this application are very good since there is hardly a manufacturing process where cleaning is not required.

2. Effect of ultrasonic energy

Early experimenters have found that ultrasonics makes possible the production of emulsions which are difficult to obtain by other means, such as oil in water, mercury in water, etc. It was found by tests that a particularly good effect was obtained when the layer of, for example, oil on water was very thin. This probably led to the well-known demonstrations of cleaning fabrics by ultrasonics.

The idea of cleaning of fabrics by means of ultrasonics was carried over to metal surfaces. The mechanism of this effect, however, is probably different. Many different kinds of solvents can be used. Oil is soluble in many of these agents. Hence it is believed that the dispersion effected by insonation accounts for the improvement in cleaning.

3. Energy, frequency, and time requirements

Tests have been conducted to determine the energy and also the time necessary for the cleaning operation. Metal parts with grease or grinding compounds were placed in the solvent bath insonated by an ultrasonic generator. In the early stages of this work, the relative intensity was determined by reading the plate current. The degree of dirt removal was observed, and rated by an observer; time was also noted. Curves were obtained of degree of cleaning vs. intensity for a constant exposure time. These curves are S-shaped. It appears from these curves that good cleaning is obtained when cavitation occurs. The energy level at which the slope of this curve changes is very close to the plate current value where intensive agitation was just observed.

These tests were conducted at frequencies of 300, 500, and 1000 kc/s. Within this range no frequency influence was observed. A magnetostrictive transducer of a frequency of 30 kc/s was used also for these tests. As expected, cavitation occurred at the surface of the transducer, and prevented energy flow into the other parts of the cleaning fluid. The cleaning effect was negligible. The level at which cavitation occurred was changed by pressurization. Again no good cleaning action was obtained as long as cavitation did not occur. Some cleaning action could be determined when the pressure was just about sufficient to suppress cavitation. Then apparently some cavitation occurred in places other than the transducer surface.

4. Application

One serious problem in industry is the removal of grinding compound from a surface. This is particularly difficult when the grinding compound has not been removed shortly after the operation but has had a chance to "cake", particularly in corners. An example in this case is the head of an electric shaver. The inner surface of the outer head has to be carefully ground to match the outer surface of the inner head. Caking of the compound occurs between block and wrapper as shown in Fig. 1. Vapour

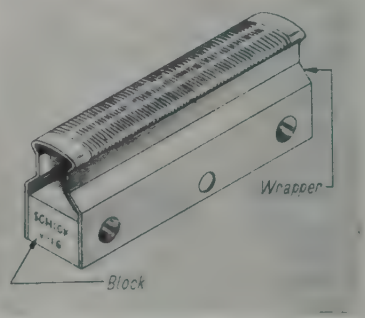


Fig. 1. Head of an electric shaver. The wrapper is tack welded to the block after being slotted and crimped to shape. The space between wrapper and block is practically zero, due to smoothness of both parts.

degreasing did not take off this deposit. Removal had been possible only by means of careful brushing. Ultrasonics removed the compound after an insonation of only 10 s.

Cleaning often limits the size of the grinding compound used. In some cases, finer compound is desired but is not used because it is so difficult to remove. Here the application of ultrasonics makes it possible to go to this finer compound and thus to reduce the surface roughness. This technique has been applied successfully to the manufacture of refrigerator compressor parts. Other applications for ultrasonic cleaning may be found in the manufacture of ball bearings, electronic tubes, cameras, electric clocks, etc.

5. Transducer

The transducer used consists of a quartz crystal, which radiates into a water or weak saline solution (Fig. 2). The energy is transmitted through the solution, and into the solvent through an acoustic window consisting of a stainless steel diaphragm 0.075 mm thick. The water or saline solution acts as one of the electrodes. It has been found that for continuous application, this type of electrode is more reliable than a metal coating directly applied to the crystal surface. It should be noticed that the acoustic window is not parallel to the crystal surface. It has been experimentally shown that a better energy transfer is thus obtained. Some work on this general problem has been published by R. D. Fay (J. acoust. Soc. Amer. **25** [1953], 623).

The crystal itself is placed in transformer oil to permit the application of the rather high voltages necessary for the operation. Details are shown in the photograph of Fig. 3. Sealing of the crystal is done by neoprene 0-rings.

For the application mentioned above, the solvent is contained in a long and narrow trough. The

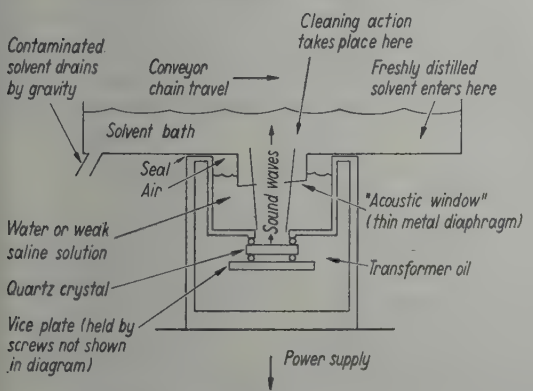


Fig. 2. Simplified schematic view of acoustic portion of the ultrasonic cleaning machine.

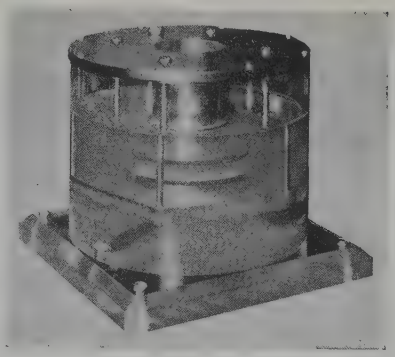


Fig. 3. Transducer.

parts to be cleaned are attached by hooks or by magnets to a conveyor belt, which dips the parts into the trough (Fig. 4). The solvent flows in the direction opposite to that of the conveyor. The solvent is connected to a distilling plant where the dirt is removed, and from which the purified fluid returns to the cleaning trough.

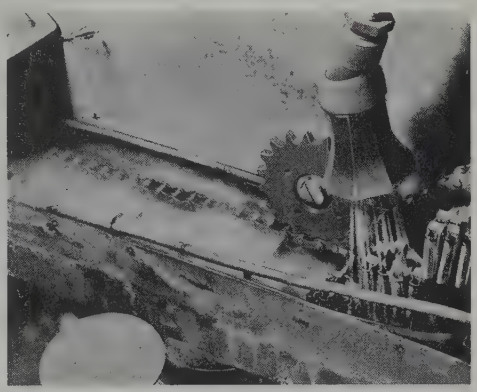


Fig. 4. Shaver heads emerging from degreasing fluid and being rinsed by more cleaning fluid, type trichlorethylene. Apparent steam is cleaning fluid vapour generated by ultrasonic vibration in the fluid.

6. Measurement of sound intensity

A problem of long standing is the measurement of the sound intensity. This is desirable to obtain an economic design of the equipment, and to assure also the passing of the parts through a field of sufficient intensity for satisfactory operation.

The method selected consists of measuring the sound intensity by observing the rate at which sound degenerates into heat in a small body of low-Q solid material. The energy is absorbed in a small pellet of laminac (polyester) in which a thermistor is imbedded. The thermistor is connected into a bridge which in turn drives an instrument such as a photo-

electric recorder. Fig. 5 shows the probe, the thermistor bridge, and a galvanometer.

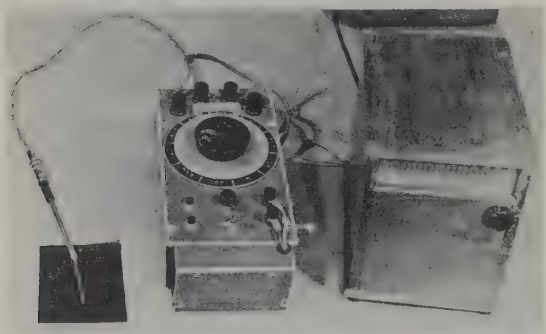


Fig. 5. The probe, the thermistor bridge and an indicating galvanometer.

7. Theory of probe

The pellet is made small and the stem is made very thin in order to disturb the sound field as little as possible. The probe diameter is 2 mm.

One may write of a plane progressive wave:

$$\frac{dQ}{dt} = KI$$

where Q = heat developed in pellet in J,
I = sound intensity in W/m²,
K = constant in m².

The heat generated within the pellet raises the temperature of the pellet above that of the liquid, and heat then flows from the pellet to the liquid.

$$\frac{dQ}{dt} = KI = MC\ \frac{dT_p}{dt} + h\,(T_p - T_l)$$

where M = effective mass of pellet in kg,
C = specific heat of the pellet in J/kg °C,
T_p = temperature of pellet in °C,
T_l = temperature of liquid in °C,
h = heat transfer coefficient in W/°C.

In the steady state case $dT_p/dt = 0$, and the temperature difference measures the intensity, according to the steady state equation:

$$KI = h\,(T_p - T_l).$$

The constant K depends on the fraction of the sound energy transferred from the liquid into the pellet and its area. The second factor does not require any comments. The first factor may be approximated.

The transfer of energy depends on the ϱc ratio of the liquid and the pellet material. This ratio is in the order of $1\frac{1}{4}$ to $1\frac{1}{2}$. The transmitted energy, hence, may be considered in the order of 95 to 97% and for all practical purposes it may be assumed that all the sound energy is transmitted. The area is, of course, πr^2.

A correction to this factor of K must, however, be made since not *all* the energy which goes into the sphere is converted into heat energy. Some of the energy is re-radiated by the vibration of the sphere. The value of K is normally determined by test but a value may be assigned to it by assuming that the ratio of re-radiated energy to heat dissipated energy is in the order of 10 to 1.

One can hence say that the proportionality constant:

$$K = A\,\eta$$

where A = effective area of pellet in m²,

$$\eta = \frac{\text{energy converted into heat}}{\text{total energy transmitted into pellet}}.$$

Hence: $K = 3 \times 10^{-7}$ m².

The value of h can be determined by experiments. When the sound source is turned off, KI equals zero and the following equation is obtained:

$$-MC\ \frac{dT_p}{dt} = h\,(T_p - T_l).$$

The sound source is turned off, and the decay slope is recorded. Since M and C are known, h can be calculated. For the pellet used $M = 4 \cdot 10^{-6}$ kg; $C = 1.25$ kJ/kg °C, h was calculated as 3 mW/ °C from a chart taken on the photoelectric recorder after the sound had been turned off.

The steady state equation may be put in a form to measure sound intensity directly:

$$I = 10\,000\ (T_p - T_l) \text{ in W/m}^2.$$

The bridge resolves differences in temperature of 0.1 °C corresponding to about 10^3 W/m². At the upper limit the pellet will maintain fair stability up to 200 °C corresponding to intensities up to $2 \cdot 10^6$ W/m² (η assumed 0.1).

In cases where the sound field is not a plane progressive wave, the reading obtained is a measurement of the total energy incident upon the pellet.

The functioning of the pellet has been checked so far in the following way. The pellet has been used to plot the field within the liquid container of the ultrasonic generator. Field plots were made at distances 4, 7 and 9 cm above the transducer surface. The results obtained are shown in Fig. 6. These figures show that the probe measures the same amount of energy flowing through these three areas. This is the result expected since it can be assumed that all the energy transmitted from the transducer into the liquid must pass through these three areas.

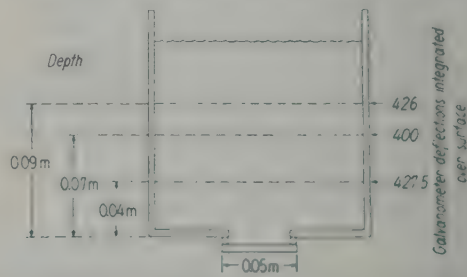


Fig. 6. Galvanometer deflections integrated over surface for three depths.

Discussion

PAHUD, J. P.: La courbe donnant le seuil d'apparition de cavitation montre que pour 1 Mc/s il faut 100 W/cm². Une telle intensité me parait dépasser de beaucoup ce qui a jamais été mesuré. Comment a-t-elle été obtenue?

LECTURER: Cavitation may occur between 0.3 and 100 W/cm² (dubious region) as shown in the paper by R. W. SAMSEL, "Physical and economic limitations in the application of sonic and ultrasonic energy to industrial processing", Amer. Instn. chem. Engng **47**, Symposium Series No. 1.

PAHUD, J. P.: Je ne comprends pas comment vous pouvez utiliser du polystyrène pour enrober le thermistor, car selon nos expériences, ce produit est détruit à des intensités inférieures.

LECTURER: Laminac is a polyester resin. It starts softening at 100 °C but it can be used for short time application under no mechanical load condition up to 200 °C.

FLORISSON, CH.: Quelle puissance globale émettez-vous dans le liquide?

LECTURER: About 3 to 4 W/cm² using a crystal of 5 cm diameter.

FLORISSON, CH.: Pourquoi n'utilisez-vous pas la méthode calorimétrique d'étalonnage plus facile?

LECTURER: We do use it at present, but the thermistor is more "industrial".

FLORISSON, CH.: Nous avons de bons résultats de nettoyage depuis environ 4 ans à $f = 18\,000$ Hz et $W_{global} = 1{,}2$ kW.

CRONSHAW, A. W.: What are the reasons for working in the "dubious" region (variation of cavitation with frequency)?

LECTURER: Cavitation is desirable for cleaning. In the "dubious" frequency region cavitation occurs throughout the liquid while at lower frequencies (non-dubious region) air bubbles appear on the surface of the transducer so preventing transmission of energy into the liquid.

CRONSHAW, A. W.: In which region is the probe used (i.e. above or below cavitation)?

LECTURER: Only below cavitation.

CRONSHAW, A. W.: Was there cavitation in the medium used for energy transfer.

LECTURER: We are here in the "dubious" region to which we have referred before.

FEDERICI, M.: How many minutes does it take to clean?

LECTURER: 3 to 10 seconds.

FEDERICI, M.: Of course machines are being employed in Europe for cleaning, but they use sonic frequencies as it seems that with ultrasonic frequencies the cloth absorbs too much power.

LECTURER: We were talking about cleaning of metal parts. Ultrasonics clean fabrics; however this application is not practical at present.

SECTION VI

ELECTRO-ACOUSTICS APPLIED TO MUSICAL INSTRUMENTS

GENERAL REVIEW

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This section of the Congress is perhaps peculiar in that it has both its objective and subjective aspects. It is not merely a question of acoustics and electricity for, as soon as one talks of "musical appreciation", physiology and psychology come in. Nevertheless, nowadays, one cannot advance far in this subject without employing electro-acoustic apparatus as will be apparent as soon as you hear individual papers in this subject, which indeed has a long history going back to PYTHAGORAS. It is my task to try to put before you the present position in this science and in doing so, to call attention particularly to the recent work of those not represented here.

Essentially our physical problems are those of mechano-acoustic and electro-acoustic transduction and radiation but the conditions under which these "transformations" of energy take place are usually far from ideal. Not only do our instruments show a varying timbre over the gamut but if played in consort may destructively interfere. On top of this the auditorium adds its acoustical pattern.

Fortunately the musical ear though very critical in some respects is tolerant in others. It is inclined to accept the best that it can hear as the absolute best or at any rate to make up the deficiency in the mind's ear. The Ancient Greeks were very clever at inventing musical scales, but I wonder if they ever heard anything like their scales actually played. A late of colleague of mine, very keen on modal music, had constructed a set of tuning forks precisely tuned in the plagal mode and assured me that that was the music the Ancient Greeks heard, but I did not believe him. Even as late as the 19th century the practice of harmony lagged far behind theory, because musical instruments were constructed on "cut and try methods" and put in the hands of inexpert players. You may remember the famous tale about CHERUBINI, faced with the absence of one of the flautists of his orchestra; when the leader said to him: "Oh, maestro, what can be worse than only one flute", CHERUBINI replied: "Two". Though the intonation of wind instruments was bad at that time few musical instrument makers would have gone as far as the primitive Red Indians who made reedpipes with side-holes in two equidistant groups of three to play up and down a scale! When asked by musicologists why they did this they are alleged to have replied: "It looks best that way and is convenient for the fingers". Even the primitive Scots have advanced beyond that stage as you will hear later in the Congress in a paper on the scale of the bagpipe.

Musicians and the music trade are cautious bodies but they are willing to give anything new a hearing witness the development of the electrophonic organ not yet universally accepted. These synthetic instruments helped in the development of electro-acoustic analysers, without which many recent advances in our knowledge of the functioning and of the means to improve the functioning of musical instruments would have been impossible.

The science of musical acoustics then falls into two parts: 1) the production and radiation of music, 2) its effect on the listener. My emphasis will be on the former since it is applied physics, pure but not simple.

One aspect of the question may properly be answered in physical terms merely, that is, how the sounds of one instrument, played solo, may be distinguished from another. This is mainly a question of timbre, but this distinction is often uncertain between two instruments at certain parts of the scale, so that, e.g. the violin steady tone is very like that of the oboe. Yet many people can distinguish a violin solo from an oboe solo.

How then does this happen? One factor is that the timbre of an instrument is not constant over the whole of its range, so that the equality just cited is unlikely to persist when the soloist passes from a note of medium pitch to one higher or lower. Another is the difference as between different instruments in the starting and build-up of steady notes, the scrape of the bow, the tonguing of the reed, etc. The first of these characteristics is called *formant*, the second *transient*, though sometimes the first word is used to cover both. The removal of the transients can make a great difference to ease of distinguishing between instruments of different or even of the same types.

The formant, in the narrower sense of the word which I prefer, is that feature in the sound of an instrument (or of a voice) which distinguishes it from another of the same breed and is largely a question of the existence of resonances which may be excited in the soundboard or box to which the primary toneproducer, vibrating string, reed, etc., is coupled. In speaking of "soundboard" one must use the word in the general sense to include all neighbouring bodies which can be set in forced vibration. For instance it ought to include the cavity resonator formed by the player's lungs, whether he is playing oboe or violin!

Another factor which must be considered is the directional characteristic. An instrument with a definite soundboard like the violin probably radiates best in the direction perpendicular to the board, a wind instrument in the direction of the bell, though this direction may be changed by local reflecting surfaces. The soundboard should vibrate as a whole up to as high a pitch as possible. If it tends to subdivide into segments — some moving out while others move in — their mutual effects at a distance will cancel each other and the sound will not radiate well.

This directivity is only, however, valid for high pitch. At low frequencies it fails and the source behaves nearly as a point source. This is a matter of some importance when one wishes to localize the sound from a particular instrument in the orchestra; for instance, a good violin, in the sense of one which if used for the solo part in a concerto will stand out from the mass of other strings. It is believed that low frequency transients help also in this respect.

A factor of similar origin which will make localisation of the music of the wind more certain than that of the strings in an orchestra is that the former are mostly playing parts which are not doubled, whereas the latter show a "chorus effect". The fact that two or more strings playing the same part can never exactly reproduce each other's effect as to loudness, pitch and timbre, and time of initiation and duration of transients (even if they were played by a machine) masks any directivity, introduces slight but rapid vibrato and smudges individual characteristics. The same is true of an organ solo stop played over chorus foundation stops on another manual. This chorus effect is, in fact, of sufficient importance to be imitated — than which nothing is easier — by the makers of electrophonic instruments.

For recording and analysing formants the usual electro-acoustic apparatus is used. It is useful to exhibit the results of octave-filter analyses at different fundamentals (steady tone) three dimensionally as Lottermoser [1] has done in his analysis of the acoustic spectra of organs.

For transients a device originally due to Dietsch and Fricke [2] is used. A portion of the trace, long enough to contain a good number of waves is converted into the "hill and dale" form and wrapped round the transparent rim of a large flywheel which is spun up to speed by an electric motor, disengaged and allowed to subside slowly to rest, while a beam of light crosses the record to fall upon a photoelectric cell connected to a vibration galvanometer. Every time the transitory speed of the wheel invokes a resonance of the galvanometer, the frequency of the component responsible can be calculated from the instantaneous speed of the wheel, measured stroboscopically (Fig. 1). A development of this apparatus is used by the British Admiralty for measuring the components in water waves.

It is impossible in the course of this lecture to deal with all musical instruments so I propose to select a few on which recent research has concentrated, since these selected researches may be taken as typical of the whole.

When one mentions violins the work of Prof. Saunders [3] and his colleagues in the U.S.A. and of Backhaus [4] and Meinel [5] in Germany at once come to mind. The former has for long been trying to improve the tone quality of violins and recently, in collaboration with Mrs. Hutchins, has



Fig. 1. A device for measuring transients.

pin-pointed some of the features which distinguish a satisfying violin. Fig. 2 shows the intensity of the fundamental tone in the notes (below) produced on several violins with the localisation of the cavity resonance (A) and belly resonances (P). They have found that the spacing of these resonances is very important. Good peaks of response near the open notes of the strings, as in the Guarnerius are important for good "singing" tone. A thinning of the edges of the wooden plates forming the belly improves the quality. According to the work of ABBOTT and PURCELL [6] and of FUKADA [7], a suitable wood for the belly or for pianoforte soundboards has a high YOUNG's modulus and a small damping factor, in which the varnish plays a part.

I have already mentioned the important part which transients play in differentiating instruments of different families. SKUDRZYK [8] opines that the old masters and their good modern equivalents radiate these transients (particularly in the low frequencies) better than less good violins, making the former stand out better in concertos. KURZ [9], on the other hand, thinks it is the "Klirrfaktor" (non-linear distortion) which is important for this differentiation and compares an old master and a mass-produced violin in this respect, finding that the former has a pronounced distortion peak about middle C.

No subject connected with the science of musical instruments has provoked greater controversy than the functioning of the flute and diapason organ pipe. The mode of coupling of the edge-tone formed at the mouth to the column of air has been of interest in aerodynamics and acoustics alike over the past fifty years. Recently there has been a revival of interest in edge tones as being a possible cause of the horrific noise put out by jet engines but the mechanism is still the subject of rival theories.

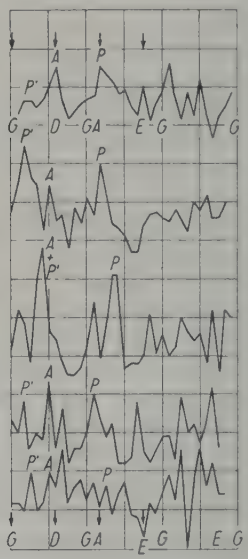


Fig. 2. The intensity of the fundamental tone in the notes (below) produced on several violins.
A: localisation of the cavity resonances,
P: localisation of the belly resonances.

When air debouches from a linear slit to fall upon the sharp edge of a metal or wooden wedge facing it, a vortex system is set up. In a way that is still rather imperfectly understood, a steady formation of vortices takes place at the slit so that the wavelength of the system — distance between successive vortices on the same side — is equal to or is a small submultiple of the separation of slit to edge.

Taking the simplest case, a vortex leaves the outer wall of the orifice as the preceding one on the same side arrives at the edge (Fig. 3a). There is a minimum slit—edge distance l_0 for any given velocity of efflux V at which a tone can be produced. The frequency of the "edge tone" is given by the frequency with which the eddies strike the wedge. If they move towards it with velocity U, and $l =$ distance between the vortices in the same row, then $f = U/l$ or since $U = aV$ where a is constant V/fl is constant.



Fig. 3. The relative positions of the vortices at different slit—edge distances.

When V is kept constant, i.e. when constant pressure is maintained behind the slit, and l is increased beyond the minimum l_0, the pitch of this "edge tone" falls in accordance with this relation until at a value approximately double of l_0, the system becomes unstable and the tone, which is now the sub-octave of the original, may suddenly rise an octave to what it was at l_0.

Experiments in which smoke is mixed with the emerging air show that the original spacing of the vortices is restored after the transition, but with twice as many between slit and edge. Fig. 3b shows their relative positions just before the transition. After the jump (Fig. 3c) l resumes its original value so that, and in agreement with the VON KÁRMÁN stability criterion, the width is halved and the original narrow vortex street is recovered. The edge must bisect the two rows of vortices if the tone is to be elicited; consequently, in the shaded area of Fig. 3c representing the space between the wide and the narrow street at the transition, only the graver tone can be elicited. In fact, after the jump has taken place, the pre-transition tone can be brought back either (1) by moving the edge into the shaded region, or (2) by pushing an obstacle from the side partly into the path of the blast and so deflecting the shaded portion of the stream on to the edge, as is done by means of a wooden "roller" in certain diapason pipes.

A number of explanations have been put forward to explain the regularity of this vortex system, but the one which seems most plausible — at any rate, for the low wind speeds used in wind instruments — is that originally due to the writer [12] who noticed a system of "secondary vortices" alongside the wedge starting at the vortex of the wedge and proceeding down the boundary layer thereof. The reaction of these secondary vortices on the phase of the air pendulation as it leaves the jet accounts for the spacing of edge tone vortices as CURLE [10] has recently pointed out. In another recent paper however POWELL [11] supports the original pressure pulse theory of WACHSMUTH.

By adopting ideas from alternating current theory the acoustician has been able to solve a number of analogous problems in sound wherever the acoustic elements could be grouped into "inertances" and "capacitances" and built up into circuits whose electrical counterparts had already been solved or were amenable to solution. In the case of an acoustic radiator whose frequency is to be varied over a certain range, the system is assumed to be driven at resonance and the damping may be neglected. In most orchestral wood-wind instruments the variation is a step-by-step one, each step being a semitone.

The problem is then usually set in the form: What adjustments of impedance of the component parts are necessary for this variation, and of the alternatives, if any, which ones will allow the player to progress with the greatest facility from one note to another? For those who are familiar with loud-speaker practice, and particularly that of the exponential horn, it is important to realise that the function of a wind instrument is not primarily to radiate sound in a specified direction, but to produce a note of frequency determined by the resonant state of the system of impedances of which it is made up. FOUCHÉ [13] has recently calculated a number of such impedance combinations. Though exponential horns have been employed on the brass-wind, they cannot be given too pronounced a flare, or their resonance peaks will overlap so much that the player is sore put to it to strike one note without warbling into another — a state of affairs which is closely approached on the modern French horn — to the anxiety of the apprentice player.

Though the notes on such an instrument as the horn are restricted to the harmonic series based on a few set speaking lengths, the wood-wind operate on a different principle. This is to make use of side-holes, whose opening and closing varies the arrangement and value of the impedance so linked in the instruments. A simple wind instrument like the flageolet is in fact the acoustical analogue of a short artificial electric line, with the side holes set as earthed branches to the main conduit. The "conductivity" (or "admittance", to use a more modern term) of these orifices can be nullified by covering with a finger or key, or reduced by half-covering, a practice adopted by flautists of the older schools. The flute then is the analogue of the cable (the air in the tube) loaded by inductances (the sideholes). Thus if all holes are covered except one, the system may be treated as a cable (up to the site of the open hole) terminated by an inductance and another length of cable in parallel, and the net impedance equated to zero for resonance. When several holes remote from the mouthpiece are open, the contribution of those lower down may be neglected, but not that of those near the first open hole. Thus the note formed by covering the holes, 1, 2, 3 and 5 may be intermediate in pitch between that fingered 1, 2, 3, 4, and that of 1, 2, 3, 4, 5.

The nature of the coupling between exciter and resonator is a matter that has interested workers in musical acoustics of late. NYBORG and his colleagues [14] are making a close study of these "jet-edge-resonator" systems which discloses the complexity of the problem, while MERCER [15] has studied the action of "voicing" by which the organ-builder adjusts his pipe to the edge tone. The

process of "voicing" by which an organ builder gets the pipe to speak as he wants it, cutting the mouth and making notches in the lower lip, is a mysterious art which I hope will be clarified in MERCER's paper.

In a flue-pipe the edge tone — which, unmodulated, would normally have a pitch proportional to the wind velocity — is brought into consonance with one of the natural modes of vibration of the column, the fundamental at moderate pressure, the overtones at higher ("overblown") pressures. Exceptionally if the pipe is "underblown", a reshuffling of the pendulations in the jet takes place whereby either the fundamental or its octave again appear. Although these underblown tones are undesirable they often appear transiently as the blowing starts. Thus in the oscillograph record of the commencement of sounding of the diapason pipe which I reproduce (Fig. 4a) the octave is apparent in the transient for about one-tenth of a second though not in the final wave-form. The build-up of pressure in this pipe lasted about one fifth of a second. Transients also occur on orchestral wind instruments when the player passes "portamento" from one note to another (Fig. 4c).

In a reed pipe the coupling between reed and column may be either tight or loose. If the reed is slim and rather inelastic because of its moist condition — like many orchestral reeds, including the brass-player's lips — it accommodates itself to whatever note the player and/or the wind pressure impose on the column. In the organ reed-pipe, however, the reed is strong and elastic and if reed and column are tightly coupled the two can only sound sympathetically when the length of the pipe is adjusted to synchronism with the reed, otherwise the system remains silent. Moreover, the transients are miniature copies of the final tone, just as in the violin (Fig. 4b).

With the partners loosely coupled, the main function of the resonator is to purify the rather raucous sound of the reed by smothering its overtones. The clarinet has interested a number of scientists because of the problems connected with the function of the reed. It seems that the reed as it vibrates in the mouthpiece closes and opens periodically a slit by which the air is admitted to the pipe.

I have mentioned the effect of material and varnish on the quality of a stringed instrument. The effect (of material) on the quality of organ pipes seems to be less important than some organ-builders suppose, for the action of the voicer can mask any

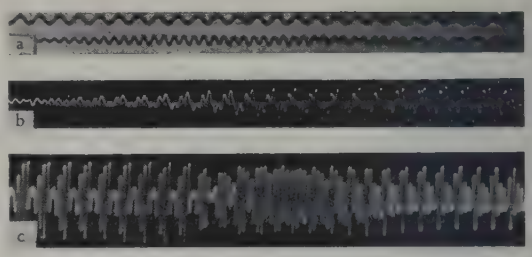


Fig. 4. a) The commencement of sounding of the diapason pipe,
b) the commencement of sounding of the violin,
c) transients occurring on orchestral wind instruments when the player passes "portamento" from one note to another.

special effect of the material. The question has been investigated by LOTTERMOSER [16] and by KNAUSS and YEAGER [17] and very recently by the well-known British organ-builder HARRISON of Durham. The latter has built four pipes to exactly the same form, including the important feature of equal wall thickness, and analysed the acoustic spectra after they have been treated equally by a skilled voicer. (I am aware that this latter "equality" is a subjective one.) Their spectra are shown in the table below.

Time does not permit a survey of all work being now done in musical acoustics, but I hope I have indicated that it offers many interesting problems of applied physics. I hope that others will be encouraged to enter this somewhat neglected field, and that in a few years time another international acoustic congress will devote some time to hearing of their work.

One subject we may then discuss is the international standard of pitch. Most of us in scientific circles adopt the recommended A = 440 c/s but, according to Prof. PASQUALINI, this standard is by no means recognised by musicians outside the national and broadcasting orchestras, nor have many tuning fork makers conformed. He suggests that we could profitably discuss the means of educating these people to adopt our recognised standard.

Table giving the frequencies f and relative sound pressures p produced by diapason pipes.

Pipe	Metal		Partials							
A	Lead and 40% tin	f	523	1093	1560	2100	2620	3170	3680	c/s
		p	60	67	25	6	10	6	—	dB
B	15% tin	f	523	1090	1570	2120	2620	3150	3730	c/s
		p	60	66	25	30	16	6	6	dB
C	50% tin	f	526	1040	1570	2100	2620	3000	3750	c/s
		p	60	53	52	80	8	4	4	dB
D	all tin	f	520	1070	1560	2080	2580	3170	3700	c/s
		p	60	45	65	25	25	5	5	dB

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NEUERE ERGEBNISSE DER ELEKTRISCHEN KLANGERZEUGUNG UND DEREN BEZIEHUNGEN ZU DER MECHANISCHEN KLANGERZEUGUNG

von R. Bierl

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Zusammenfassung. Die elektronischen Klanggeneratoren haben wegen ihrer rein elektrischen Schwingungserzeugung gegenüber den elektromechanischen den Vorzug einer wesentlich größeren Vielfalt an Ausdrucksmöglichkeiten. Dadurch kann der Spieler die den Klang definierenden Größen: Tonhöhe, Tonstärke und Klangfarbe zugunsten einer Erweiterung des musikalischen Vortrages zu einem größeren Anteil entsprechend seinem Willen beeinflussen. Die Ordnung der Klangfarben in einem Intervallzirkel für die Frequenzkurven zunächst zweiwelliger Gebilde zusammen mit dem durch Verzerrungs- und Unsymmetriegrad angegebenen Erregungsspektrum ist nicht nur für diese neuartigen Instrumente wichtig, sondern kann auch die musikalisch-akustische Forschung der herkömmlichen Instrumente fördern.

Summary. Improved means of expression than by electro-mechanical generators are offered by electronic sound generators (without moving parts) allowing the player to change his pitch, loudness and timbre and so to widen his musical means of expression. The arrangement of timbres along a closed circle of intervals representing the frequency response curves of the system of vibrations (for instance coupled circuits) as well as some features of the spectrum of oscillations exciting the system (viz. the degree of distortion and asymmetry of the oscillations) is not only important for these new types of instruments but can also assist musical-acoustical research on the old established types.

Sommaire. Les instruments de musique électronique offrent beaucoup plus de possibilités d'expression que les instruments électromécaniques. Les oscillations étant produites de façon électrique, la partie des grandeurs définissant un son musical (hauteur, intensité et timbre) dont dispose l'exécutant peut être accrue pour élargir les possibilités musicales. La représentation des timbres en un cercle fermé d'intervalles, déterminés par les courbes de réponse du système de vibrations (circuits à couplage par exemple) et le spectre des oscillations excitant le système (le spectre lui-même étant conditionné par le taux de distorsion et de dissymétrie), est non seulement importante pour ces nouveaux types d'instruments, mais peut également présenter de l'intérêt pour les recherches d'acoustique musicale aux instruments usuels.

1. Einleitung

Musikinstrumente — und ganz allgemein die Klanggeneratoren — mit elektrischer Schwingungserzeugung können auf einem der elektromechanischen oder auf einem elektronischen Verfahren beruhen. Zu den ersteren sind alle Anordnungen zu rechnen, bei denen auf galvanischem, dielektrischem, elektromagnetischem oder lichtelektrischem Wege eine Abtastung von mechanischen Bewegungen (Auslenkungen) oder ihren zeitlichen Differentialquotienten stattfindet. Dabei kann entweder — wie z.B. im Falle der Generatoren mit rotierenden Zahnscheiben — ein Profil durch einen relativ zu diesem, meistens gleichförmig bewegten Spalt oder auch die Bewegung von schwingenden Körpern abgetastet werden. Als den Spaltmethoden verwandt kann man auch die Verfahren ansehen, die von Schallspeichern Gebrauch machen; die „musique concrète" geht dabei von natürlichen Schallereignissen aus; in anderen Fällen (Meyer-Epplersche Klangmodelle) werden Aufzeichnungen elektronischer Schwingungsgeneratoren verwendet.

Die bereits erwähnten elektronischen Verfahren arbeiten mit rein elektrischer Schwingungserzeugung, also ohne mechanisch bewegte Teile.

Geht man von der Forderung nach einer Möglichkeit des unmittelbaren, augenblicklichen Musizierens aus, wobei die mehr konstruktiven Verfahren mittels Schallspeicherung ausscheiden, dann sind, vor allem bei Berücksichtigung der im Laufe von einigen Jahrhunderten entwickelten musikalischen Disziplinen, Problemstellung und -lösung ähnlich wie bei den mechanischen Klangerzeugern (Musikinstrumenten, Schlagzeugen und Sprache). Zwischen den einzelnen Teilfragen der Erzeugung und Beeinflussung der Schwingungen läßt sich also eine sehr weitgehende Analogie aufstellen [1], da lediglich anstelle mechanischer Schwingungen elektrische vorliegen, die dann aber immer durch einen Lautsprecher in Schall umgeformt werden.

Gegenüber den herkömmlichen Musikinstrumenten, deren Charakter durch die Grenzen des Tonhöhenumfangs, des Frequenzbereichs der abgestrahlten Schallenergie usw. gegeben ist — Grenzen, die je nach dem Grad der Spieltechnik jedoch eine außerordentlich große Vielfalt von Klangwirkungen einschließen, so daß wirklich neue kaum mehr möglich sind *) —, fällt vor allem bei elektronischen Generatoren diese Begrenzung weg oder wird zumindest stark vermindert, so daß dem Spieler in dieser Richtung freiere Wahl überlassen bleibt. Das darf aber natürlich nicht so verstanden werden, daß die herkömmlichen Bereiche bestehender Instrumente erweitert werden sollen, denn dabei ginge ja ihre Definition verloren. Da aber auch die

*) Mixtureffekte sind gleichbedeutend mit Mehrstimmigkeit und — siehe Orgelbau — auch bei mechanischen Klanggeneratoren möglich.

Imitation mechanischer Instrumente von heute oder einer früheren Zeit weder exakt möglich noch erwünscht ist — wegen der Analogie mechanischer und elektrischer Schwingungsgebilde werden allerdings in der Vielzahl der möglichen Fälle auch solche enthalten sein die angenähert den herkömmlichen ähnlich sind —, erhebt sich die Frage nach dem Sinn solcher Neuentwicklungen. Dieser kann nun darin gesehen werden, daß die elektronischen Instrumente vor allem mit ihren in größerem Maße veränderbaren charakteristischen Grenzen dem Musiker zusätzliche Möglichkeiten für Komposition und Vortrag bieten. Deren Ausnützung setzt jedoch neben einer ausreichend schnellen und sicheren Wählbarkeit der folgenden charakteristischen Größen eine Zuordnung dieser Größen in einem absolut gültigen Klangfarbenordnungssystem voraus.

2. Die charakteristischen Größen und ihre Realisierung

Jeder Klang ist bekanntlich definiert durch seine (Grund-)Frequenz (Tonhöhe), Amplitude (Tonstärke) und seine Kurvenform (Klangfarbe); musikalisch verwertbar ist er erst, wenn jede dieser drei Größen mindestens zu einem gewissen Betrag veränderbar ist. Andrerseits scheinen die Grenzen, innerhalb derer solche Veränderungen möglich sind, für die herkömmlichen Klanggeneratoren charakteristisch zu sein, während die je nach dem Willen oder Können des Spielers mehr oder weniger vorliegende Ausnützung der Variationsbereiche für den persönlichen Vortrag wesentlich ist.

a) *Tonhöhe.* Selbst bei herkömmlichen Instrumenten mit fester Intervallteilung kann man die Möglichkeit von Abweichungen von den festen Frequenzstufen feststellen; zum Teil sind sie periodisch schwankend (Vibrato), gelegentlich aber auch nichtperiodisch mit sehr kleiner Änderungsgeschwindigkeit (absichtliche kleine Verstimmungen).

Von dem — in der Regel in festen (Halbton-) Stufen intonierten — Spielbereich der Musikinstrumente mit elektrischer Schwingungserzeugung wird — vielleicht auf Grund gewisser geschichtlicher Tatsachen — verlangt, daß er — möglichst mehrmals um je eine Oktave — für die Zwecke der Reinstimmung und die aus musikalischen Gründen notwendige willkürliche periodische oder nichtperiodische Verstimmung jeweils getrennt und unabhängig voneinander um kleine Beträge stetig versetzt werden kann. Diese Forderungen sind, wie man leicht sieht, gleichzeitig nur von einem obertonarmen Generator zu erfüllen; als zweckmäßig hat sich die Gegentakt-Rückkopplungs-Schaltung in besonderer Ausführungsform erwiesen, weil hier die Thomsonsche Resonanzkreisformel weitgehend, auch mit der erforderlichen Genauigkeit von $0{,}5^0/_{00}$, gültig bzw. beherrschbar bleibt. Ein besonderes Problem liegt natürlich auch darin, daß die *Intonation* weder bei einer *Oktavversetzung* noch bei den *Verstimmungen* bzw. *Stimmungskorrekturen* gestört werden darf.

b) Der mittlere Verlauf der *Tonstärke*, die Lautstärke, ist durch verhältnismäßig einfache Mittel einstellbar. Anders ist es mit ihren Augenblickswerten, deren zeitlicher Verlauf auch wieder zum Teil instrumentgegeben (*Ansprache*) oder willkürlich (*Artikulation*) ist. An sich ist es wohl naheliegend, ähnlich wie z.B. bei den Blasinstrumenten, Lautstärke und Artikulation mit ein- und demselben Organ darzustellen; schwieriger ist das Problem aber zweifellos bei dem instrumentgegebenen Anteil, der Ansprache. Infolge der für jede Tonhöhe verschiedenen Resonanzeigenschaften des betreffenden Schwingungssystems ist es zweckmäßig, die daraus resultierenden Ausgleichsvorgänge, soweit sie für Grundton und Obertöne gleich oder ähnlich sind, durch *Pseudoausgleichsvorgänge* (Verstärkungsregelung durch exponentiell abklingende Gleichspannung) zu ersetzen, den Restanteil dagegen in den Filtern zu erzeugen (vgl. weiter unten).

c) *Klangfarbe.* An sich kann man natürlich jede gewünschte periodische obertonhaltige Schwingung sowohl durch *Addition sinusförmiger Schwingungen*, als auch durch den *Abbau einer obertonreichen Schwingung* mittels Filter herstellen. Abgesehen davon, daß die Addition von vielen Harmonischen etwa bis zur Ordnungszahl 20 erhebliche technische Schwierigkeiten (Aufwand) mit sich bringt, scheint aber dieses Verfahren auch nur Klänge zu liefern, die als starr empfunden und daher für die üblichen Zwecke abgelehnt werden.

Jeder herkömmliche Klanggenerator hat nun — mindestens in seinem Schallstrahler — ein Übertragungssystem, das durch seine oft sehr verwickelte Frequenzkurve einen entscheidenden Einfluß auf die resultierende Kurvenform hat. Sowohl bei einem Vergleich von verschiedenen herkömmlichen Instrumenten, als auch an dem Verlauf der Klangfarbe während eines musikalischen Vortrages auf einem Instrument kann man aber erkennen, daß das Spektrum der obertonhaltigen Schwingung außer durch die erwähnte Systemfrequenzkurve auch noch an anderer Stelle, nämlich meistens im eigentlichen Schwingungsgenerator, entweder beeinflußt werden kann oder vorgegeben wird. Es ist daher zweckmäßig, sowohl von einem eventuell in Grenzen variierbaren *Erregungsspektrum*, das also dem Willen des Spielers unterliegt, auszugehen, als auch eine Änderung desselben durch die *System-Frequenzkurve* — als instrumentgegebenen Anteil — anzunehmen.

Mit einem Erregungsspektrum, das durch *Verzerrungsgrad* — Obertongehalt — und *Unsymmetrie-*

grad — Amplitudenverhältnis der ungeraden und geraden Harmonischen — charakterisiert ist, wobei natürlich jede dieser Größen von der Tonhöhe abhängen kann, und mit einer mehr oder weniger komplizierten, d.h. mehrwelligen, Frequenzkurve dürfte man praktisch jede beliebige Klangfarbe mehr oder weniger angenähert darstellen können; hierbei ist dann auch die zweifellos sehr wichtige Abhängigkeit der Kurvenform von der Tonhöhe [2] berücksichtigt. Im Falle nichtperiodischer Erregungsschwingungen kommen, wie noch bemerkt sei, Geräusche ähnlich z.B. denen der Schlagzeuge zustande.

Es zeigte sich bald, daß mit den sogenannten einwelligen Schwingungsgebilden (an elektronischen Instrumenten kann das besonders gut demonstriert werden) keine befriedigenden Effekte erreicht werden können. Bei zweiwelligen ist die Zahl der möglichen Schaltungen schon ziemlich groß; eine systematische Betrachtung ergibt aber, daß die möglichen Fälle in die auch bei den einwelligen Gebilden auftretenden Gruppen: Tiefpaß (TP), Hochpaß (HP), Bandpaß (BP) und Bandsperre (BS) eingeteilt werden können. Die einzelnen Frequenzkurven unterscheiden sich dadurch, daß die Hauptmerkmale — Durchlaßbereich unter-, oberhalb einer Grenzfrequenz oder innerhalb eines Frequenzbandes bzw. Sperrbereich in diesem — durch zusätzliche Unregelmäßigkeiten verändert sind.

Schon bei dem — gegenüber den mechanischen Instrumenten — sehr vereinfachten zweiwelligen Fall ergibt sich also eine so große Zahl von Parametern — Typ, Hauptfrequenz, Kopplungsgrad, Dämpfungsdekrement —, daß die Zuordnung von handlichen Spielhilfen — mit zum Teil mehreren Stufen — bereits einige Schwierigkeiten bereitet; hier wird sich im Laufe von längeren praktischen Versuchen der Musiker herausstellen müssen, wieviele und welche von den genannten Parametern häufig und evtl. sogar kontinuierlich verändert werden müssen, da ja wohl kaum alle in gleicher Weise für die eingangs erwähnte Klangfarbenmodifikation wirksam sind. Es mag daran erinnert werden, daß bei der Sprache in erster Linie Kopplungsgrad und Hauptfrequenz verändert werden.

3. Das Klangfarbenordnungssystem

Die Gruppierung der möglichen Frequenzkurven auch bei mehrwelligen Gebilden in die vier genannten Hauptgruppen verleitet zu einer Orientierung, zusammen mit gewissen Zwischenfällen, zu einem geschlossenen Kreis. Alle möglichen Schaltungen, auch die Entartungen, lassen sich recht vernünftig in einem solchen Diagram unterbringen [1]. Eine solche Darstellung könnte *Klangfarben-Intervall-Zirkel* genannt und ähnlich wie der Quintenzirkel der Harmonielehre angewandt werden und hätte außerdem den Vorteil der absoluten Gültigkeit. Möglicherweise könnte ein solcher Zirkel auch bei der allgemeinen musikalisch-akustischen Forschung an herkömmlichen Instrumenten Fortschritte ermöglichen.

Schrifttum

[1] Bierl, R., Über die Beziehungen zwischen elektrischer und mechanischer Klangerzeugung. Z. angew. Phys. 5 [1953], 231.
[2] Saunders, F. A., J. acoust. Soc. Amer. 18 [1946], 395; vgl. Anm. [1].

Diskussion

Meyer-Eppler, W.: Liefert die Einführung von Bandsperren klangliche Ergebnisse, die sich von den mit breitbandigem Durchlaßbereich erzielbaren sehr unterscheiden?

Vortragender: Ja, wenn entweder die Bandbreite des Sperrbereichs groß oder die Dämpfung innerhalb des Sperrbereichs groß ist. In dem vorgeführten Gerät werden Bandsperren der ersteren Art mit einer Bandbreite von mehr als einer Oktave bevorzugt.

ELECTRONIC TUNERS FOR MUSICAL PITCH

by J. J. Geluk
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Summary. The stability of electronic oscillators has surpassed the accuracy of mechanical tuning devices; especially in respect to temperature variations the electronic oscillators can be given almost a completely stable frequency. Even with great limitation in circuit parts and costs, the constancy of the frequency can be made better than that of a steel tuning fork. Nowadays simple electronic devices can be made according to the international standard of musical pitch.

Sommaire. La stabilité des oscillateurs électroniques est devenue supérieure à la précision des dispositifs mécaniques d'accord; en particulier, on peut réaliser des oscillateurs électroniques dont la fréquence est quasi absolument stable vis à vis des variations de la température. Même s'il y a des conditions très strictes quant aux éléments du circuit et au prix de revient, la fréquence peut être rendue plus constante que celle d'un diapason en acier. On peut, à l'heure actuelle, réaliser des dispositifs électroniques simples qui sont conformes au diapason international standard.

Zusammenfassung. Die Stabilität elektronischer Schwingungserzeuger übertrifft in der Genauigkeit die zum Stimmen benutzten mechanischen Systeme; insbesondere im Hinblick auf Temperaturschwankungen kann man die Frequenz des elektronischen Gerätes völlig stabil halten. Selbst bei großen Einschränkungen hinsichtlich der Bauteile und Kosten kann eine bessere Konstanz erreicht werden als bei einer stählernen Stimmgabel. Man kann heute einfache elektronische Geräte herstellen, die der internationalen Norm für die musikalische Tonhöhe entsprechen.

1. Introduction

After several centuries of gradual change of the tuning note A, the conference of London in 1939 adopted unanimously a frequency of 440 c/s in preference to the figures of 435, 439 and others. Besides the argument that 440 was no prime number and in the neighbourhood of the most commonly used frequency of 439, it gave merit to the tendency of increasing steadily the tuning tone.

It was expected then that a limit was set to this upwards movement, but to assure this, convenient and adequate apparatus should be at the disposal of the musicians.

Hitherto, the small steel tuning fork had fulfilled this task, because it gave the best precision in the easiest way; nevertheless the temperature did change the frequency and what was worse, one could alter its length by elementary means, so changing the frequency. For some instruments, e.g. the piano, the tuning fork showed the same sign of frequency drift with temperature, other instruments however gave opposite sign. To offer a real standard of the musical tuning tone, the frequency was determined without taking the temperature into account.

It is obvious that, strictly speaking, fixed instruments never hold this standard to a very close approximation and an additional recommendation should be given to manufacturers of musical instruments, so that during performance under "normal" conditions the tuning is as correct as possible.

At the time the standard was agreed upon, it was technically possible to manufacture sub-standard oscillators, whose frequency was practically independent of temperature, but the question arose, how an orchestra had to work and tune with these modern electronic apparatus.

If the apparatus only gave the fundamental tone, there came many complaints of the musicians, who were not accustomed to tune to such "pure" tones and soon the sound of the oboe was simulated by electronic means.

Even the best physical approximation could not overcome all the complaints of the musicians and in many cases the method had to be abandoned.

In the Netherlands the Standardization Committee therefore decided to emphasize the physical way of tuning by recommending to observe the rhythm of the beats, thus abandoning tuning by "pitch". Following this procedure for one instrument only (e.g. oboe), the orchestra is free to tune by pitch to this "reference" instrument. This procedure proved to be adequate.

With electronic tuners giving a complex note it was found, that if e.g. an oboe and a violin tuned separately to the sub-standard, the instruments were out of pitch playing together. Evidently the instruments tune to pitch rather than to beat notes and that is essentially what matters. For instruments with fixed tuning, one should keep the standard tone as close as possible at the atmospheric conditions under which the instrument is normally used. There is electronic apparatus on the market which gives a complete equitempered octave, with reference to the piano and analogous instruments. But a piano-tuner never holds this tuning exactly for all octaves, but expands his gamut towards higher octaves, in order to let the instrument sound "brilliant". This again is an argument for abandoning the idea of simulating the oboe; obviously this implies, economically and in respect to size of the reference-standard, an advantage. Also the power output of the generators can be made low, as only one musician should be able to hear it; on the other hand it must not be too low, because often the tuning of this reference instrument must be done in a concert-hall, where the noise-level is very high due to the audience and the other musicians.

2. Principles of electronic tuners

The generator, as source of the apparatus, should be insensitive to temperature fluctuations, resulting in a short warm-up period to give the constant frequency. A mechanical or electrical resonance-circuit is used, having a certain quality factor Q which may be as high as 3000 or as low as $\frac{1}{3}$. In both cases amplification increases this value to sustained oscillations and some means for limiting the amplitude should be set. There is no need to avoid any possible distortion and, if modern temperature-dependent resistances (N.T.C., "Kaltleiter") are used, the limitation is sufficient. Special attention must be paid to the influence on the generated frequency of the amplifier; especially for the low Q systems it is necessary to stabilize the amplifier by introducing sufficient feedback. The only variation in frequency that remains then is due to the atmospheric influence on the resonance circuit itself.

The most commonly used circuits are: (a) crystal, (b) tuning-fork, (c) L-C, (d) R-C.

Before deciding what circuit is most suitable, we should consider what is necessary for the constancy of the frequency. With tuning forks, having a decay-time of about 10 s ($Q = 2000$) less than two beats during the decay cannot be detected, so it is of no use to have a greater accuracy than $\pm$ 0.2 c/s or $\pm 5 \cdot 10^{-4}$. If the temperature be the only reason of frequency-shift, the temperature-coefficient should be less than 10^{-4} per degree centigrade, if we tolerate a working temperature range of $\pm$ 5 °C.

A crystal circuit therefore, giving a temperature-coefficient of 10^{-6}/°C without controlling its own temperature, is more than is needed; therefore this circuit with its rather complicated dividing networks is unsatisfactory for this purpose.

The tuning fork of normal steel has a temperature-coefficient of approx. 10^{-4}/°C and is therefore just adequate; but additionally, these forks are pressure-dependent ($3 \cdot 10^{-6}$ per cm Hg).

An improvement can be found by taking a special alloy and sealing the fork in a closed chamber; then instability of about $0.5 \cdot 10^{-4}$ of the standard frequency can be achieved, but with considerable increase of cost and loss of simplicity. Due to the high Q value, the rather long time of building up the oscillations remains as a disadvantage of such mechanical resonance circuits.

An electronic tuner, using this circuit, is schematically shown in Fig. 1. Separate odd and even harmonics can be added to the fundamental. A special circuit cancels the fundamental tone in the odd harmonic series, obtained by clipping the sine wave in tube B_3. The even harmonics are produced by tube B_4, which works as a full-wave rectifier. Both series are de-emphasized for the lower numbers of the harmonics, giving good contributions in the 12 kc/s range. As all the three attenuators of the apparatus add a different frequency spectrum, the total is the sum of the partial spectra.

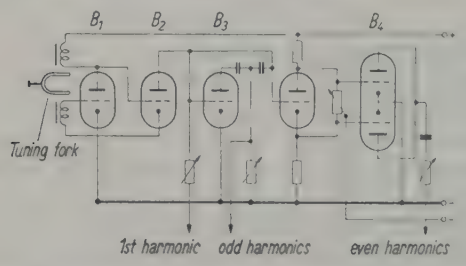


Fig. 1. Electronic tuning equipment with a tuning-fork; harmonics can be added at will.

It was found that the most accepted tone-quality had important harmonics up to the 22nd (9680 c/s); its spectrum, acoustically measured, was as indicated in Fig. 2.

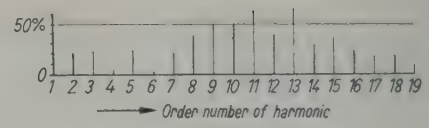


Fig. 2. Acoustical spectrum of oboe imitation. Strength of the harmonics relative to fundamental (= 100%).

In other countries the same experiment was done in a different way; the findings were however different from those, given in Fig. 2. In Fig. 3 a comparison is given between the results of the B.B.C., the R.T.I. at Nürnberg and our own measurement of the spectrum of a real oboe.

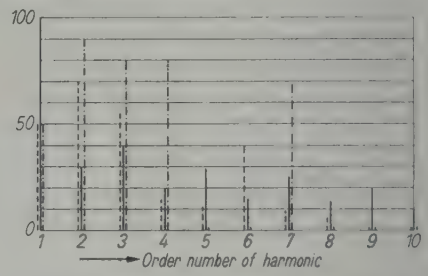


Fig. 3. Comparison of the spectra of oboe imitations of the B.B.C. (———) and the R.T.I. (—·—), and the spectrum of a real oboe (own measurement) (- - - - -).

After the unsatisfactory results, obtained with such tones, as described in the introduction, two small electronic tuners were made giving an almost pure tone of 440 c/s. One of these has a normal steel tuning fork as resonance circuit, while a second type was made with an R-C frequency determining circuit. By suitable choice of resistances with positive and negative temperature coefficients, it has proved to be half as much dependent on temperature-influences as a steel fork ($0.5 \cdot 10^{-4}$/°C). The circuit is given in Fig. 4, in which R_6 is a N.T.C. resistor, regulating the amplitude of the oscillation. The frequency-determining components $R_{1,2,3}$ and $C_{2,3}$ are all temperature-compensated.

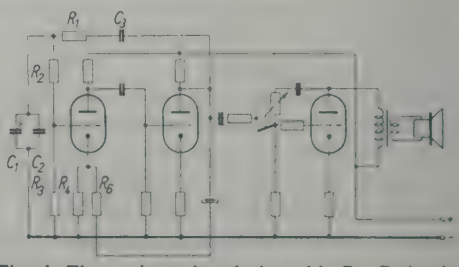


Fig. 4. Electronic tuning device with R - C circuit.

To avoid long warming-up periods, the volume-control is combined with a switch in the last amplifier stage.

To obtain good stability over very long periods resistances however are less reliable; in this respect

L-C circuits are appreciably better and commercial electronic tuners apply mostly this circuit. Also the temperature coefficients of the elements have a reduced influence on the frequency. In Fig. 5 a circuit is given, giving also a simple possibility to introduce higher harmonics; after 15 minutes warming-up period the frequency is guaranteed to be 440 c/s $\pm$ 0.2%.

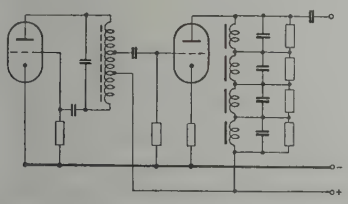


Fig. 5. Commercial electronic tuning device with L - C circuit.

A series L-C circuit is used in another commercial tuner, acting as a selective filter in the feedback loop of an amplifier. Switching the capacity, the frequency is changed by factors of $\sqrt{2}$, while six values of the self-inductance can be chosen to match the standard tone or slightly different values.

In Fig. 6 the main circuit is given. The regulation is obtained by means of a resistance with a positive temperature coefficient in the cathode circuit of the first amplifier.

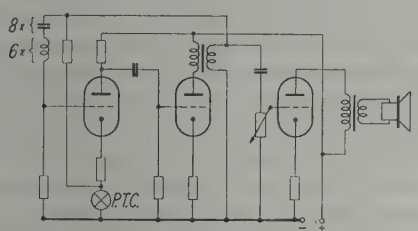


Fig. 6. Commercial electronic tuning device with a series L - C circuit.

Interesting is the fact that this instrument gives the octave of the standard tone (880 c/s), having the advantage of better radiation by a small loudspeaker and giving higher loudness level with the same sound pressure.

Tuning by observation of beat-tones then has the additional advantage that these are doubled in rhythm and therefore more easily detectable. The condition must be fulfilled that the tuning instrument has reasonable harmonic content in its sound.

3. Conclusion

Nowadays it is possible to manufacture small electronic tuners, giving a reference signal for tuning-purposes of satisfactory accuracy. Although mechanical resonance circuits such as tuning-forks are adequate, electrical components even can give better results. Imitating the sound of an oboe is not necessary and even sometimes confuses the aim of standardisation. An inquiry, recently undertaken by the U.E.R. (Union Européenne de Radiodiffusion) gives a good report of how the standardization makes progress as far as Europe is concerned. About 80% of the European countries use the standard pitch of 440 c/s, while only one small country (Morocco) has adopted a standard of 435 c/s. Further two small countries have no standard in current use.

From the point of view of standardization, it is sufficient to introduce to musicians a reference-tone, which allows them to compare the actual pitch they have chosen and also to maintain this pitch. The latter is important in broadcasting practice, where several parts of a musical item are to be recorded at different times.

Comparison should always be done by the observation of beat notes. For instrument manufacturers it is important that the instruments should exhibit their best properties when tuned to the normal 440 c/s frequency.

Finally it seems to be of no use to insist that the musicians keep to the standard A, because for the musical performances only the musicians themselves are responsible; however, technicians can supply them with simple apparatus which enables them to determine objectively the deviations they think to be justified.

Literature: see page 262

Discussion

YOUNG, R. W.: Does the portable standard just described to produce the twelve chromatic notes, produce the correct frequencies within 0.2 c/s?

LECTURER: The differences between the nominal values and the measured frequencies are small. Taking the 440 c/s tone as "Kammerton", we found the whole octave correct within $\pm$ 0.5 c/s (in respect to the frequencies from 523 to 1047 c/s. For other values of the "standard" A somewhat larger discrepancies were found but never exceeded 0.25%.

EIJK, J. VAN DEN: Why is it necessary to tune with an accuracy of 0.2 c/s? The accuracy of playing music will be much less than this I think.

LECTURER: Of course the accuracy of the standard must be one order better than the highest accuracy ever wanted for an instrument. Moreover well-tuned instruments are very desirable in extreme cases, such as string instruments playing unison on free strings or instruments with a fixed tuning playing together (organ, piano etc.).

TRAUTWEIN, F.: Welche Abhängigkeit besteht bei den beschriebenen R-C- und L-C-Generatoren von der Netzspannung?

LECTURER: Sie ist gering; eine genaue Messung dieser Abhängigkeit ist nicht einfach, weil auch noch Temperaturschwankungen hinzukommen. Insgesamt haben wir gemessen, daß 10% Netzspannungserniedrigung nur 0,2‰ Frequenzänderung zur Folge hatte.

A SOUND REVERSAL TECHNIQUE APPLIED TO THE STUDY OF TONE QUALITY

by W. H. George

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Summary. Reversal of magnetic-tape sound-recordings is used to investigate the influence upon tone-quality of growth and decay of sounds. If the instantaneous overtone structure (as technically defined in the paper) changes with time, reversal of the sound profoundly alters the tone-quality. The idea of tone-quality as the overtone structure of the "steady state" of a note modified by "transients", is abandoned for the idea of tone-quality as a "pattern property".

Sommaire. On inverse des enregistrements sonores sur ruban magnétique pour étudier l'influence de l'établissement et de l'amortissement des sons sur le timbre. Si la structure instantanée en harmoniques (telle qu'elle est définie au point de vue technique dans l'article) varie avec le temps, l'inversion du son modifie profondément le timbre. On abandonne le concept de timbre basé sur la structure en harmoniques de "l'état permanent" ("steady state") d'une note modifié par les "transitoires" ("transients") et on le remplace par le concept de timbre considéré comme un "diagramme de configuration" ("pattern property").

Zusammenfassung. Umkehrungen von Magnettonaufzeichnungen werden benutzt, um den Einfluß des An- und Abklingens von Schallvorgängen auf die Klangfarbe zu untersuchen. Wenn das Obertonverhältnis sich zeitlich ändert, wird die Klangfarbe durch eine Umkehrung grundlegend verändert. Man mußte von der Auffassung, die Klangfarbe als Obertonverhältnis, das von Einschwingvorgängen („transients") modifiziert wird, zu erklären, abgehen und sie als eine Gestalt („pattern-property") ansehen.

The present study of tone-quality belongs to a class of acoustical research of which the results can be published in sound only. Sound recording is an essential part of the research, but the recordings are not a substitute for actual sounds. Some of the sounds heard from the recorder could not be produced "live" and they must be heard to be believed.

There is general agreement that *tone quality* (*timbre, Klangfarbe*) is that which enables a hearer to distinguish, by hearing, between notes of the same pitch and loudness played upon two different instruments. The differences may be as small as those between two violins, or between two voices speaking the same vowel. The standard method of studying these differences for musical, speech and other sounds has been to make an harmonic, or periodic analysis of the sound produced. The initial and final parts of a sound, as well as diminuendo, crescendo, vibrato and transients in general are regarded as modifications of the "steady state".

For convenience and brevity in this paper, the term *ios* at time t will be used to represent *instantaneous overtone structure* defined as follows. For a sound with harmonic overtones and total duration T, let time be reckoned zero at the beginning of the sound under analysis. Starting at time t move forward in time by an interval δt equal to the period of the lowest-pitched component in the sound. Let the variations occurring during δt be repeated infinitely in the positive and negative directions of time. Then the harmonic analysis of the resultant is here defined as the *ios* or *instantaneous overtone structure at* time t. (For inharmonic sounds the definition must be modified to make δt equal to the lowest common multiple of all the components or, alternatively, a phase component must be included and periodic analysis used instead of harmonic analysis [1]). Using this definition it may be said that in the bulk of published research on tone-quality it is assumed that (a) tone quality is largely determined by *ios* and (b) *ios* is independent of t/T. The present research began, not with a study of tone-quality, but with a crucial experiment designed to test the sustaining power of the modern piano. Because the generation of the sound takes place in a minute fraction (of the order 0.0001) of the apparent duration of the sound it has been suggested that the sustaining effect is largely psychological. To test this, a simple magnetic-tape recorder (kindly lent by the Westminster Hospital, London in 1950) was used to record and reproduce musical passages so that the individual notes kept their places in the score but their sounds were individually reversed. The experiment proved that the sustaining effect was genuine. But the surprising result was that although *ios* was unchanged *none of the sounds could be recognised as made by a piano.* It is well known that for the piano *ios* is a function of t/T.

With the generous co-operation of E.M.I. Ltd., the experiments were repeated with high-fidelity recording and reproducing apparatus and a first-rate concert grand piano. The resulting complete change of tone-quality was confirmed and the results were demonstrated at a meeting of the Acoustics Group of the Physical Society (February 22nd, 1951).

In an improved form they are here reproduced from material prepared by the British Broadcasting Corporation for broadcast talks.

Sound illustration 1. Chopin, Prelude in C minor Op. 28 No. 20. In some bars the notes were individually reversed. In other bars the notes were played normally so that the distortion of the recording and reproducing system including the acoustics of the room could be judged.

As the reversed piano tones lack the usual precision of attack the listener cannot be sure when each reversed note begins. In the next two recordings this difficulty was overcome by arranging that the left-hand part was played normally so as to define the time pulse, whilst the notes of the right-hand part were reversed.

Sound illustration 2. Opening of Chopin's Berceuse Op. 57. The notes of the right-hand part were individually reversed, whilst those of the left-hand were played normally.

The third recording was prepared to show the influence of tempo on the sustaining power of the piano which, at the tempo used, appears to be complete.

Sound illustration 3. Opening of Chopin's Study in F minor (number 25). This consists of two independent parts, the right hand playing regular groups of three single notes, each individually reversed, whilst the left hand plays regular groups of four single notes, each played normally.

These two records, made in 1952 and not yet broadcast, represent a remarkable feat of engineering skill by the B.B.C. Engineers. Entirely independent recording units had to be used for each part. One part had to be reversed and the two accurately synchronised. Some months later the writer's attention was directed to American experiments, carried out with a player piano, on synchronisation in piano playing [2]. According to this work, the accuracy of illustrations 2 and 3 would appear to be of the order 0.01 s. Tape speeds of 0.75 m/s (30 inch/s) were used. As records 2 and 3 include both normal and reversed sounds, electronic, transducer and room-acoustic distortions are the same for both, and the finally observed complete change of tone-quality of sounds of the piano is established.

Experiments on other sounds have been carried out only with simple tape recorders using tape speeds of 0.19 or 0.38 m/s (7.5 or 15 inch/s).

The results may be summarised as follows:

a) No *complete* sound (made by non-electrical means) of duration T sounds the same on reversal, although reversal does not alter the *ios*.

b) If the "steady state" (as observed on an oscilloscope) of any sound be reversed, its sound is unchanged but, in general, the original source of the sound cannot be identified with certainty even though its *ios* is that given in the literature as characteristic of that sound.

c) If any other portion of a complete sound, for which *ios* varies with time, is clipped from the original and played separately, it sounds quite different in the forward and backward directions. In neither reproduction is it recognisable.

Furthermore it may be noted that complete sounds acquire additional qualities depending on their environment e.g. (a) any single note played on any instrument successively as the tonic or the leading note or (b) the same vowel sound placed in different words.

The writer attaches particular importance to the technique used in illustrations 1 ··· 3. Highly critical observation is essential to research in tone-quality. The observer must know exactly when the observed phenomenon is to occur. By using musical passages and accurate synchronisation the observer is given this information. Recordings played successively cannot be compared with the same degree of accuracy. Unfortunately the technique is beyond the resources of the ordinary laboratory. Four recorders and two highly skilled engineers were used at one stage of the experiments.

In the present state of knowledge the writer has given up the idea of tone-quality as something associated with a "steady state" modified by "transient". Instead he regards it as a "pattern" property [3].

The sound illustrations are part of a series prepared for talks to be given in the B.B.C. radio programme "Music Magazine". The writer wishes to thank the Westminster Hospital for the loan of a simple tape recorder in 1950, Messrs. E.M.I. for high fidelity recordings made in 1951 and the British Broadcasting Corporation for permission to use some of their sound recordings (not yet broadcast at the time of the Congress) and for kindly recording a spoken version of this paper.

References

[1] This type of definition, in terms of the action to be taken by the experimenter, is discussed in George, W. H., The scientist in action. A scientific study of his methods. Emerson Book Inc. New York 1938, pp 96—97 and 285.

[2] Vernon, L. N., Synchronisation of chords in artistic piano playing. Univ. of Iowa studies 4 [1936], 309.

[3] George, W. H., ref. [1] Chapt. VII. The term is defined on pp. 120—121.

ÜBER DIE KLÄNGE VON GLOCKEN UND ORGELN

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Zusammenfassung. Es werden die elektroakustischen Meßmethoden beschrieben, die in der Physikalisch-Technischen Bundesanstalt, Braunschweig, angewandt werden, um ein objektives Bild vom Klangcharakter einer Glocke oder einer Orgel zu erhalten.

Summary. The electro-acoustical methods used in the Physikalisch-Technische Bundesanstalt, Braunschweig, for getting objective data of the character of a bell or an organ are described.

Sommaire. On décrit les méthodes électroacoustiques employées au Physikalisch-Technische Bundesanstalt, Braunschweig, pour caractériser objectivement le timbre d'une cloche ou d'un orgue.

In dem letzten Kriege sind in Deutschland wie schon im ersten Weltkrieg zahlreiche wertvolle Glockengeläute und Orgeln verlorengegangen. Die Wiederbeschaffung dieser für das kirchliche Leben so bedeutungsvollen Musikinstrumente in gleicher oder möglichst besserer Qualität hat die Frage aufgeworfen, wodurch eigentlich diese Qualität physikalisch bestimmt sei. Wir haben uns aus diesem Grunde in der Physikalisch-Technischen Bundesanstalt in Braunschweig bemüht, sowohl für die Glocken wie auch für die Orgeln einfache, die bekannten elektroakustischen Meßmethoden verwendende Verfahren zu entwickeln, die ein möglichst objektives Bild von der Klangqualität der Instrumente zu geben vermögen **). Von meinen Mitarbeitern hat sich dabei Herr STAPPENBECK in erster Linie der Glocken angenommen, während Herr Dr. LOTTERMOSER sich den Orgeln gewidmet hat.

Wenn wir mit dem *Glockenklang* beginnen, so ist die erste wichtige Frage die nach der *Frequenz* der einzelnen Teiltöne. Diese Teiltöne aus der Form und der Massenverteilung der Glocke zu berechnen, ist ziemlich aussichtslos; sie einzeln jedoch elektrisch anzuregen, gelingt leicht mit Hilfe eines guten *RC*-Tongenerators und eines elektromagnetischen Anregungssystems, das an geeigneter Stelle auf die Glockenwand aufgesetzt wird. Die Glockenschwingung wird zur Frequenzmessung von einem auf die schwingende Glocke aufgesetzten Körperschallempfänger wieder abgenommen und einer Braunschen Röhre zusammen mit der Anregungsspannung zugeführt. Man verändert solange die Frequenz der Anregungsspannung, bis eine Ellipse als Lissajous-Figur entsteht. Entfernt man den Anreger von der Glocke und läßt diese ausklingen, so muß während des ganzen Ausklingvorganges die Ellipsenform erhalten bleiben, wenn man sicher sein will, wirklich die Eigenfrequenz der gerade angeregten Glockenschwingung zu messen. Die Frequenzmessung geschieht wieder durch Beobachtung der LISSAJOUS-Figur mit dem gleichen Oszillographen, dem jetzt eine Normalfrequenz von 1000 Hz zugeleitet wird. Bei einer gewissen Übung ist eine Meßgenauigkeit von 1/10 Hz für Teiltöne bis zur 40. Ordnung leicht zu erreichen.

Die *Amplituden* der Teiltöne interessieren in ihrem relativen Verhältnis zueinander beim normalen Anschlag der Glocke. Zu ihrer Bestimmung nehmen wir den Anschlag einer Glocke auf ein Magnetophonband auf und machen eine Suchtonanalyse dieses Schallvorganges für das Zeitintervall einer 1/5 s unmittelbar nach dem Anschlag. Um eine Analyse für ein so kurzes Zeitintervall ausführen zu können, muß der Bandausschnitt, der die Länge von etwa 15 cm hat, vervielfältigt werden, und es müssen mehrere gleichartige Stücke zu einem endlosen Band der Gesamtlänge von etwa 1 m zusammengeklebt werden. Die Analyse kann dann mit beliebiger Genauigkeit, meist mit einer Analysierschärfe von 8 Hz, mit einem automatisch registrierenden Suchtonanalysator vorgenommen werden. Abb. 1 zeigt eine Analyse eines Glockenklanges kurz nach dem Anschlag. Wir erkennen das typische Bild einer Reihe verschieden starker nicht harmonischer Teiltöne. In Deutschland bezeichnet man die Teiltöne beginnend mit dem tiefsten Ton als Unteroktave (1), Prim (2), Terz (3), Quinte (4), Oberoktave (5) usw.

Allen Teiltönen ist eine bestimmte *Schwingungsform* der Glocke zuzuordnen. Diese Schwingungsform kann durch Abtasten des Glockenkörpers bei sinusförmiger elektrischer Anregung ermittelt werden. Für den vorliegenden Fall ist dies bis zum 17. Oberton geschehen. Die erhaltenen Schwingungsbilder sind oben in der Abb. 1 durch Skizzen, die die Knotenlinien in Draufsicht auf die Glocke veranschaulichen, dargestellt. Die angeschriebenen Zahlen beziehen sich jeweils auf die Ordnungszahlen

*) Nach Untersuchungen mit W. LOTTERMOSER und H. STAPPENBECK.

**) Die umfangreichen Glockenuntersuchtungen, insbesondere an Stahlglocken, wurden durch den *Bochumer Verein für Gußstahlfabrikation* ermöglicht.

der Teiltöne im darunter gelegenen Spektrum. Wir finden, wie bekannt, eine nach hohen Frequenzen rasch zunehmende Unterteilung der schwingenden Glockenfläche.

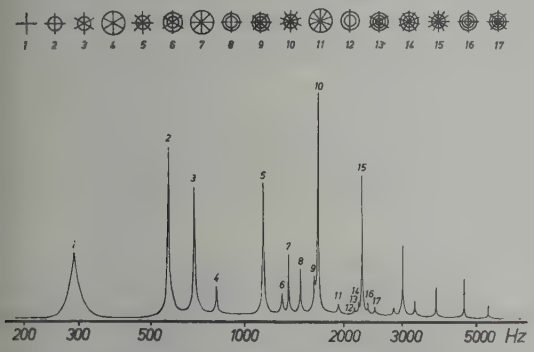


Abb. 1. Spektrum und Schwingungsform einer Glocke unmittelbar nach dem Anschlag. Die Teiltöne mit nur einem Knotenring (Lauttöner) 2, 3, 5, 10, 15 usw. dominieren.

In einer anderen Darstellung kann man einen noch besseren Überblick über die zusammengehörigen Teiltöne erhalten. In Abb. 2 sind auf der Abszisse die Frequenzen der Teiltöne, auf der Ordinate die Anzahl der Knotenmeridiane, also der Knotenlinien, die von der Krone zur Mündung der Glocke verlaufen, aufgetragen. Wir sehen mehrere zusammengehörige Kurven, die sich jeweils durch die Zahl der Knotenringe, das sind die waagerecht um die Glocke herumlaufenden Knotenkreise, unterscheiden.

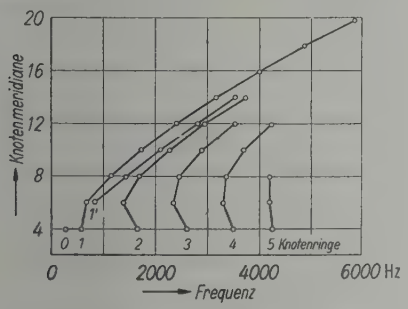


Abb. 2. Die Frequenzlage der Teiltöne einer Glocke in Abhängigkeit von der Anzahl der Knotenringe. Die oberste Kurve ist die Kurve der Töne mit einem Knotenring (Lauttöner).

Die oberste bis zu sehr hohen Frequenzen zu verfolgende Kurve enthält alle Teiltöne, die eine ansteigende Anzahl von Knotenmeridianen bei immer nur einem Knotenring besitzen. Es zeigte sich, daß gerade diesen Tönen im Glockenklang eine besondere Bedeutung zukommt, da sie durchweg die Teiltöne mit großer Lautstärke stellen. Wir haben die Tonfolge daher die Reihe der *„Lauttöner"* genannt und wollen rückblickend ihr Auftreten am Beispiel der Abb. 1 kurz verfolgen. Es gehören hier zu den Lauttönern: Die Prim (2), die Terz (3), die Oberoktave (5), die Quinte zur Oberoktave (10), die Doppeloktave (15) und die noch höher gelegene Quarte, Sexte und Tripeloktave. Man erkennt deutlich, daß alle diese Töne wesentlich über die anderen Teiltöne herausragen und somit im Glockenklang dominieren. Diese Erkenntnis ist dann von großer Wichtigkeit, wenn man darangeht, durch Änderung der Glockenform die Teiltöne so zu legen, daß sie möglichst als reine Oktaven, Terzen und Quinten im Spektrum erscheinen. Hier kommt der richtigen Ausrichtung gerade der Kurve der Lauttöner die größte Bedeutung zu. Leider muß ich mir aus Zeitgründen versagen, auf dieses interessante Gebiet näher einzugehen.

Eine Möglichkeit, wenigstens den Intensitätsschwerpunkt des Spektrums etwas zu verschieben, sei jedoch kurz angedeutet. In gewissen Grenzen kann man das *Klöppelgewicht* und auch die Höhe des Aufhängepunktes des Klöppels variieren. Man erhält so z.B. bei Vergrößerung der Klöppelmasse, wobei man die Relativgeschwindigkeit Klöppel — Glocke entsprechend verringert, ein stärkeres Hervortreten der tiefen Teiltöne gegenüber den hohen, was sowohl theoretisch begründet als auch durch Analyse des Klangspektrums bewiesen worden ist.

Auch auf das *Ausklingen* der einzelnen Teiltöne kann ich nur kurz eingehen. Die Vermutung, die man oft hört, daß das langsamere Ausklingen der Bronzeglocken gegenüber den Stahlglocken auf die in Bronze geringere innere Dämpfung zurückzuführen sei, ist nicht richtig. Die innere Schwingungsdämpfung in den Glocken ist immer klein gegenüber der akustischen Strahlungsdämpfung und nur auf diese geht die kürzere Abklingzeit der Stahlglocken zurück. Manche Stahlglocken besitzen gegenüber Bronzeglocken bei gleicher musikalischer Tonhöhe und gleichem Gewicht eine fast doppelt so große Oberfläche. Die Strahlungsdämpfung dieser Stahlglocken ist daher doppelt so groß und die Abklingzeit der Unteroktave z.B. etwa halb so groß wie die vergleichbarer Bronzeglocken.

Auch die oben näher erläuterten Lauttöner weisen wegen ihrer besonderen Schwingungsform besondere Strahlungs- und Abklingeigenschaften auf. Die Abb. 3 zeigt die Amplitudenverteilung längs des Glockenmantels bei Anregung der Lauttöner. Man erkennt, daß bei hohen Frequenzen die wirklich strahlende Fläche immer kleiner wird, was die gleichfalls längeren Abklingzeiten erklärt, die man für höhere Lauttöner im Gegensatz zu den in ihrer Nähe liegenden Teiltönen anderer Schwingungsformen findet. Natürlich sind die Abklingzeiten nur relativ länger. Da wir annehmen können, daß bei

jeder Energieumsetzung in einer Schwingungsperiode etwa der gleiche Energieverlust eintritt, werden die absoluten Ausklingzeiten etwa proportional zur Frequenz kleiner, d.h. die tiefen Töne leben bei weitem am längsten.

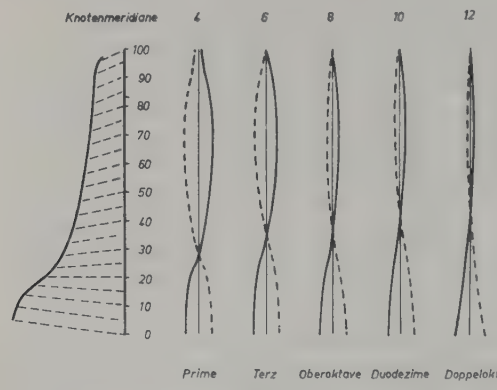


Abb. 3. Schwingungsamplitude der Glockenwand bei Teiltönen mit einem Knotenring (Lauttöner).

Zum Schluß dieses Abschnittes über den Glockenklang noch ein paar Worte über den *Schlagton*. Er ist zweifellos kein reeller Ton, sondern wird bei der Wahrnehmung des Gesamtglockenklanges vom Ohr gebildet. Vieles deutet daraufhin, daß die in der Nähe der Prim sich häufenden Differenztöne erster und zweiter Ordnung, die auf die vielen Oktaven und Quinten im Glockenspektrum zurückgehen, dort gewissermaßen einen „Tonhöhenschwerpunkt" erzeugen, der vom Ohr als Schlagton empfunden wird.

Wenn ich nun zum *Orgelklang* übergehe, so muß ich vorausschicken, daß ich auch hier in der kurzen Zeit wieder nur das Wichtigste und auch dieses nur in grober Vereinfachung bringen kann. Zu eingehenderem Studium möchte ich auf die ausführliche Arbeit von W. Lottermoser in der Acustica [1] hinweisen.

Es hat sich gezeigt, daß für einen Überblick über den Klangcharakter einer Orgel eine Oktavsiebanalyse des Plenums ausreicht. Das Plenum enthält die wichtigsten Register einer Orgel, nämlich Prinzipale und Mixturen, und kann zur Charakterisierung des Klanges dienen. Die Aufnahme der Klänge geschieht auf der Empore in etwa 15 m Abstand vom Pfeifenwerk. Ist ein solcher Platz nicht vorhanden, wird man einen ähnlichen, z.B. im Kirchenschiff in der Nähe der Kanzel, wählen.

Um die stehenden Wellen im Raum hinreichend ausgleichen zu können, genügt es, nach einem Vorschlag von E. Thienhaus, jeweils drei nebeneinander liegende Tasten (drei temperierte Halbtöne) gleichzeitig anzuschlagen. Diese Klänge werden über acht Oktavfilter von 50 ··· 10 000 Hz analysiert und zwar durchgehend für die gesamte Klaviatur. Zeichnet man alle so erhaltenen Spektren perspektivisch, den jeweils angeschlagenen Tasten zugeordnet hintereinander auf, so bekommt man einen guten Überblick über den Klang des gesamten Orgelwerkes.

In der Abb. 4 sind solche Spektren von 6 Orgeln wiedergegeben. Wir finden links zunächst die Spektren dreier berühmter im Original erhaltener Barockorgeln aus der Zeit um 1750 (Ottobeuren, Erbauer Riepp; Weingarten, Erbauer Gabler; Rot a. d. Rot, Erbauer Holzhay), rechts daneben die Klangbilder moderner Orgeln, wie wir sie in ähnlicher Qualität alle kennen.

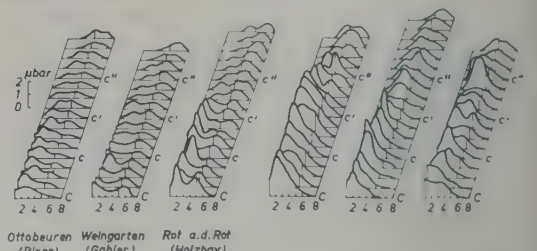


Abb. 4. Klangspektren von Orgeln;
links: drei Barockorgeln (um 1750),
rechts: drei moderne Orgeln (um 1910).

Die Unterschiede zwischen den alten und modernen Orgeln sind auffallend und lassen sich etwa folgendermaßen formulieren:

1. Die von modernen Orgeln abgegebenen Schallenergien sind größer.
2. Die Spektren der alten Orgeln zeigen eine fast gleichmäßige Energieverteilung über den gesamten Hörbereich, bei den modernen Orgeln werden die tiefen Teiltöne ausgesprochen bevorzugt.
3. Bei den alten Orgeln weist das Spektrum zwei Gipfel auf. Der zweite Gipfel verschiebt sich meist nach höheren Tönen vom A-Formant zum I-Formant. Das Spektrum der modernen Orgel besitzt nur einen Gipfel bei tiefen Frequenzen, im übrigen ist der spektrale Verlauf recht unregelmäßig.

Auch hinsichtlich der *Einschwingvorgänge* unterscheiden sich die alten und die neuen Orgeln. Eingehende oszillographische Untersuchungen der Einschwingvorgänge von Plenumklängen haben gezeigt, daß diese Klänge im Falle der Verwendung von Tonkanzellen merklich schneller anklingen, als bei Verwendung von Registerkanzellen. Dies beweist die große Bedeutung der *Tonkanzelle*, die offenbar die gegenseitige Mitnahme der einzelnen Pfeifen ermöglicht, was die Registerkanzelle nicht tut. Da bei schnellem Einschwingen der Pfeifen eine wesentlich präzisere Tongebung möglich ist, sollte

man in diesem Punkt möglichst auf dies alte Bauprinzip wieder zurückkommen.

In Deutschland hat der Klang der Barockorgel schon zahlreiche Freunde gefunden, und es scheint so, als wenn man in Zukunft immer mehr danach streben wird, alte Barockorgeln nach dem Originalklangbild wieder herzustellen, darüberhinaus aber auch neue Orgeln nach diesem Klangideal zu bauen. Bei diesem Bestreben wird es sicher von Nutzen sein, sich zur laufenden objektiven Kontrolle des Erreichten der hier beschriebenen elektroakustischen Meßverfahren zu bedienen.

Schrifttum

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EXPERIMENTAL STUDY OF TUNING PROBLEMS

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Summary. The equal temperament in music is a compromise, accepted in a period not possessing electronic equipment. Organ music especially will recover rich harmony when played with perfect intervals. Methods are described by which this can be done.

Sommaire. Le tempérament égal, en musique, constitue un compromis qu'on a dû accepter quand on ne disposait pas d'instruments électroniques. La musique d'orgue, en particulier, recouvrera une harmonie riche si elle est jouée avec des intervalles parfaits. On indique les moyens d'arriver à ce résultat.

Zusammenfassung. Die gleichschwebende Stimmung in der Musik ist ein Kompromiß, der in einer Zeit entstanden ist, als es noch keine elektronischen Geräte gab. Besonders die Orgelmusik würde beim Spiel mit reinen Intervallen wieder harmonisch reicher werden. Es werden Methoden angegeben, mit denen man dieses erreichen kann.

Ancient organs had only keys in one plane, as can be seen from old pictures; they supported the church singing. The tones produced were the Pythagorean ones, the seven tones of the scale only related by harmonic fifths (3/2) and octaves (2/1). The intervals can be written in powers of 2 and 3, forming a one-dimensional chain of increasing powers of 3:

$$\begin{array}{ccccccc} \frac{2^2}{3} & \frac{1}{1} & \frac{3}{2} & \frac{3^2}{2^3} & \frac{3^3}{2^4} & \frac{3^4}{2^6} & \frac{3^5}{2^7} \\ F & C & G & D & A & E & B \end{array}$$

This scale has only one size of whole tone: $9/8 = 170$ milli-octaves (abbrev. mo) and also only one size of semitone: $256/243 = 75$ mo. This milli-octave, $2^{0.001}$, is a practical unit, the octave being the only unequivocal interval of value 2/1.

In the middle ages harmony entered as a new element in the musical art and the tone scale was adjusted with the harmonic third, proposed by ZARLINO. The harmonic third has the value 5/4, the Pythagorean third 81/64. The intervals can now be written as powers of 2, 3 and 5 in a two-dimensional field of increasing powers of 3 and 5 (Fig. 1).

As can be seen, the tones A, E and B of PYTHAGORAS are lowered by 80/81, or a comma. There exist in this scale two kinds of whole tones: C — D = 9/8 = 170 mo and D — E = 10/9 = 152 mo; the difference is the comma 80/81 = 18 mo. The semitone is E — F = 16/15 = 93 mo. The harmonic third is C — E = 5/4 = 322 mo and the harmonic fifth is C — G = 3/2 = 585 mo. Modulation to another key, for instance from C to G, removes the rhombus in the chart one place to the right over the tonefield, giving other values for the intervals of some tones.

This tone system has three major and three minor triads:

$$\begin{array}{ccccccccc} & & A & & E & & B & & 5 \\ & & | & & | & & | & = & | \\ & & F{-}C & & C{-}G & & G{-}D & & 4{-}6 \\ D{-}A & & A{-}E & & E{-}B & & & 1/6{-}1/4 & 10{-}15 \\ | & & | & & | & = & & | = & | \\ F & & C & & G & & & 1/5 & 12 \end{array}$$

Such a system is perfectly harmonic, but has the great disadvantage of having two values for the second C — D, as a consequence of the two values for D lying a comma (81/80) apart.

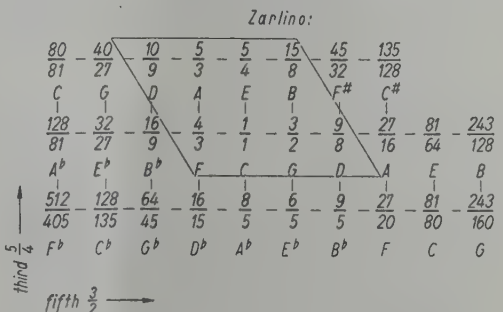


Fig. 1. The fifths-thirds related tonefield.

In order to avoid two keys for D on the key-board the fifth may be taken 1/4 comma smaller than corresponds to the ratio 3/2 (see Fig. 1), yielding a system with slightly imperfect fifths (581 instead of 585 mo) and still perfect thirds (322 mo). This

system contains a D that divides the interval of the two D's of ZARLINO's system in the middle and is called therefore the *mean tone system*. It is the system used for organ practice before BACH. This system too is not cyclic, i.e. one arrives at different values for G♯ and A♭, D♯ and E♭, etc. if the sequence of fifths is extended upward and downward. The two components of each of these pairs of tones differ 34 mo from each other. The system is nearly cyclic however and can therefore adequately be represented by a spiral (compare G♯ and A♭, D♯ and E♭ etc.).

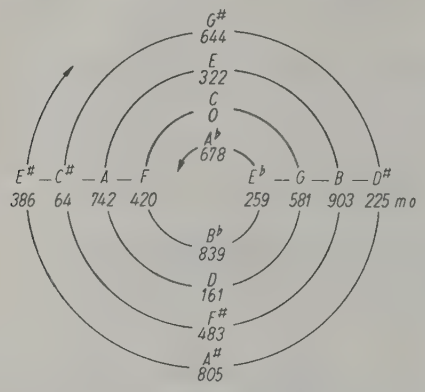


Fig. 2. The mean-tone spiral.

Modulation to another key on a mean tone tuned instrument means choosing another set of 7 tones more inward or outward on the spiral. So modulation facilities on a normal instrument with twelve keys pro octave are restricted to a few closely related keys.

From the above it is easily seen that full freedom of modulation can only be obtained in one of the following ways:

a) by increasing the number of keys without limit if the mean tone system is adhered to, or by making provisions for changing if necessary the frequency of some keys by means of stops; this is impracticable, for the instrument builder as well as for the musician;

b) by forcing the spiral into a circle by extending the fifths again from 581 to 583 mo, thus making the two components of the pairs G♯ and A♭, etc. to coincide exactly. This leads to thirds of 333 mo (and fifths of 583 mo) and to poor harmony. This system is the well-known equal temperament of to-day with 12 equal semitones per octave;

c) by subdividing the octave into 31 or 53 equal intervals, leading to almost perfect thirds, fifths and sevenths. The key-boards of such instruments become very complicated however and make it nearly impossible to play the existing organ music;

d) by using normal equal-tempered tuning, but providing for possible adjustment of intervals during the performance. A major third should be narrowed by raising the lower component by 6 mo, while flattening the higher component by the same amount, thus making the interval 322 instead of 333 mo. An equal but opposite adjustment should be made for the minor third. These alterations in frequency can be effected automatically.

In order to estimate in a subjective way what discrepancies of perfect tuning are tolerable and to determine which system of tuning is preferred by musicians and listeners, we have built an experimental electrophonic organ, comprising twelve identical rotating electro-magnetic generator sets, each driven by a synchronous motor and adjustable in frequency over about 85 mo plus and minus the nominal value, each supplying one complete tone family: C, C♯, D, ··· B. Since the twelve generator sets can be accurately adjusted to a desired pitch by means of twelve very stable *RC*-generators, mastering the synchronous motors, all systems of tuning can be realised. It is even possible to change the tuning during a performance, thus following the musician in his modulations. So it is a unique instrument, e.g., for playing, say, in the mean tone system without getting into trouble while modulating.

The preliminary conclusion is that normal listeners do not hear any difference between music played in equal or in mean tone temperament. The musicians do hear the difference and prefer the mean tone temperament and a fortiori the just intonation, but only in broad terminating chords and for choral-like music. They refuse to accept new duties in handling pitch adjusters, neither do they like the fluctuations in pitch, caused by instantaneously corrected thirds.

The equal temperament seems to be a good compromise, not only for physical urgencies, but for musical reasons too.

AN ACOUSTICAL STUDY OF THE HIGHLAND BAGPIPE

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Summary. The frequencies and intervals of the notes in the bagpipe scale have been measured by the use of a double beam oscillograph, in conjunction with a valve-maintained tuning fork.

Sommaire. On a mesuré, au moyen d'un oscillographe à deux faisceaux et d'un diapason entretenu par tube à vide, les fréquences et intervalles des notes émises par la cornemuse écossaise.

Zusammenfassung. Die Frequenzen und Intervalle der Töne der Dudelsack-Tonleiter wurden mit Hilfe eines Zweistrahloszillographen in Verbindung mit einer elektrisch erregten Stimmgabel gemessen.

1. Introduction

The bagpipe is one of the oldest forms of musical instrument, having been played since remote times in all the countries of Europe. Although the instrument in its native, and more or less original, form is still played in many countries, it is usually associated with Scotland and with the Highlands of Scotland in particular. The form of the instrument which was developed there has become the most popular throughout the world; it has been adopted by all the English-speaking nations and has in some countries superseded the native form.

It is apparent to the critical listener that the scale of the Highland bagpipe conforms neither to just intonation nor to equal temperament. In view of the popularity of the instrument and the wide variety of music which can be produced from its restricted range, the scale is of considerable interest to musicians and to physicists. The first object of the present study was to attempt a definition of the frequencies and intervals of the notes making up the bagpipe scale.

2. The instrument

The Highland bagpipe has as its principal component a chanter, from which nine notes can be produced. The flow of air to the chanter is maintained and regulated by a blowpipe fitted with a valve and by a bag acting as an air reservoir. From the bag project three drones, two tenor and one bass, which supply an unvarying harmony.

Near the air inlet to the chanter is a reed, a double vibrator made of Spanish cane bound to a copper staple. The characteristics of the note it produces vary greatly with the temperature and humidity of the air supply.

The reed used in the drones is a piece of cane in which a single beating tongue has been cut; this is one of the oldest methods of maintaining a musical note.

The chanter is made by boring two conical holes along the axis of a wooden rod, about 37 cm in length. The apex of the longer cone is upward (near the air inlet) and the apex of the shorter cone points downward. The cones overlap to give the region of narrowest bore, about 7 mm in diameter. The chanter has a series of seven holes along its length, as well as the thumb-hole, nearest to the reed and on the opposite side of the pipe to the other seven. Two cross-holes near the lower end of the chanter are bored at right angles to the others.

The drones are cylindrical tubes, with some parts widened to permit of tuning.

3. Previous work

Several attempts have been made to measure the frequencies of the notes of the chanter — usually by pipers who had little scientific training or by physicists who were dependent on others to tell them when a reasonably accurate pipe scale was being played. A. J. ELLIS [1] (1885) made some tests in which the frequencies were estimated by comparison with tuning forks. The piper was an Englishman and an amateur and the pipe was accordingly not likely to be accurately tuned. A survey of eight chanters was made by G. E. ALLAN [2](1940) with some success, though again the pipers were not of the first rank. The method of measurement was to tune a monochord to each note in turn — a lengthy procedure which made conditions difficult for the piper.

4. Apparatus

For a more accurate determination of the scale, the chanter was played in front of a moving-coil microphone connected, through an amplifier, to one beam of a double-beam oscilloscope. A valve-maintained standard tuning fork and a beat-frequency oscillator were connected to the X and Y plates respectively of a second oscilloscope. By maintaining a chosen LISSAJOUS figure on the screen of this instrument, the output of the oscillator could be kept at a convenient frequency with an accuracy of 1 in 10^5. This output was also fed to the second beam of the double-beam oscilloscope. With the time base disconnected, records of the two wave

forms were obtained with film moving at about 65 cm/s in a camera attached to the screen. The notes of the scale were played, sometimes in order of frequency and sometimes in common musical phrases; the time taken to play the nine notes varied between 3 and 6 seconds. The frequencies were calculated by comparing the two traces on the developed film.

5. Results

Eighteen chanters were tested and the results, using the usual convention, in which the lowest note of the bagpipe scale is denoted G and the highest A′, were as follows:

note	G	A	B	C	D	E	F	G′	A′	
mean frequency	410	459	516	573	620	690	766	826	914	c/s

The intervals between successive notes were calculated, in cents, for each chanter. The mean values of the intervals are given below, with those reported by ELLIS and ALLAN for comparison.

	mean intervals in cents		
note	ELLIS	ALLAN	present work
G			
	191	197	199
A			
	197	192	196
B			
	145	182	187
C			
	154	155	133
D			
	208	181	189
E			
	150	194	182
F			
	156	163	134
G′			
	191	151	184
A′			

The notes produced by the drones were also examined. For a chanter with A = 455, the tenor drones when tuned gave a fundamental of frequency 227 c/s with a strong harmonic at a frequency of 455 c/s. The bass drone gave 114 c/s as its fundamental and 342 c/s as its predominant harmonic.

6. Conclusions

a) The frequency of the note called A by pipers was found in this investigation to be 459.3 c/s, with a standard deviation of 5.6 c/s. This is nearer to A sharp (467) than to A (440) by present standards. Considerable benefit could be derived from the adoption of a standard pitch for bagpipes; we suggest A = 459. The difference between this figure and the frequency of the orchestral A or A sharp is not important since, on account of the difference in the construction of the scales, bagpipes cannot easily play in unison with other orchestral instruments. The deviation from A = 459 among the pipers who assisted us in this investigation is quite small, and is no more than that commonly found among players of other instruments when accurate measurements are made.

b) It appears to us that the bagpipe scale is made up of relatively simple intervals, though it is different from the scale of any other musical instrument. From the measurements now reported, we suggest that the bagpipe scale is made up of the following intervals:

note	suggested ratio	interval cents	observed interval cents
A			
	9/8	204	196
B			
	10/9	182	187
C			
	27/25	133	134
D			
	10/9	182	189
E			
	10/9	182	182
F			
	27/25	133	131
G′			
	10/9	182	184
A′			

A fuller formulation and discussion of this proposal is being prepared for publication elsewhere.

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Discussion

RADEMAKERS, A.: The intensity seems to fall steeply at the octave note. Is this normal or only a peculiarity of this special instrument?

LECTURER: The falling-off in amplitude at the top of the scale is always observed in the Highland bagpipes. It occurs also in the Irish bagpipes.

ZUR STIMMUNG DER MUSIKINSTRUMENTE

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Zusammenfassung. Es wird über Normstimmtonmessungen, den Bau eines elektro-optischen Stimmgerätes für Musikinstrumente und über damit erzielte Ergebnisse berichtet. Die Hörgenauigkeit bester Stimmer ist erstaunlich hoch.

Summary. An account is given concerning measurements of the normal musical pitch, the construction of an electro-optical tuning apparatus for musical instruments and the results obtained with it. The auditory accuracy of the best tuners is astonishingly high.

Sommaire. On donne les résultats des mesures du diapason normal, et on décrit un dispositif électro-optique d'accord des instruments de musique, en indiquant ce qu'il permet d'obtenir. La précision de l'ouïe des accordeurs est étonnamment grande.

1. Nichteinhaltung des Normstimmtones

Unsere Messungen an Orchestern, Orgeln, Klavieren, Blasinstrumenten und Akkordeons zeigen in Übereinstimmung mit Ergebnissen anderer [1], [2], daß eine Einhaltung des Normstimmtones $a^1 = 440$ Hz heute kaum stärker angestrebt wird als vor der Londoner Akustikkonferenz 1939. Wir nehmen deshalb im Rahmen unserer Forschungs- und Prüfstelle auch Einfluß auf den Musikinstrumentenbau selbst. Mittels Gutachtergremien aus hervorragenden Instrumentenbauern und Künstlern prüfen wir in Musterstücken unsere industrielle Produktion an wichtigeren Musikinstrumenten handwerklich und klanglich und, soweit die wissenschaftlichen Grundlagen bereits vorliegen, auch meßtechnisch.

Im Rahmen der Prüfungsbestimmungen werden auch höchstzulässige Fehlergrenzen für die Einhaltung des Normstimmtones festgelegt. Sie sind den Eigenschaften der Instrumentengattung angepaßt; denn es wäre kaum zweckmäßig, vom Musikinstrumentenbau auch dann die Einhaltung sehr kleiner Fehlergrenzen zu verlangen, wenn die Tonhöhe stark von den Erregungsbedingungen abhängt.

Tabelle I zeigt für den eingespielten Zustand (ohne den Erwärmungseinfluß) in Mittelwerten die bei Blasinstrumenten große Abhängigkeit der Tonhöhe von den Anblasebedingungen, ihren Ziehbereich. Selbst die Oboe, das Standardinstrument der Orchester für die Abgabe des Stimmtones, ist in ihrer Tonhöhe stark veränderlich, auch im eingespielten Zustand.

Tabelle I
Ziehbereich der Tonhöhe bei Blasinstrumenten

Instrument	Ziehbereich in Cents
Böhmflöte	68
Klarinette	44
Oboe	34
Fagott	70
Saxophon	42
Blockflöte	84
Trompete	36
Waldhorn	48
Posaune	28
Akkordeon	12
Mundharmonika	20

Es ist allein schon auf Grund des großen Ziehbereiches der Tonhöhe und des Erwärmungseinflusses nicht verwunderlich, wenn Orchester sehr verschieden hoch musizieren, und es dürfte dringend notwendig sein, frühere Vorschläge zur Anwendung von Normstimmtongebern [3] weitgehend zu verwirklichen. Es gibt bereits eine Reihe guter Normstimmtongeber [4], [5], [6] und Herr GELUK berichtete über weitere. Wir selbst haben einen von Temperatur und Netzspannung weitgehend unabhängigen brückenstabilisierten elektronischen Normstimmtongeber hoher Genauigkeit gebaut. Ich glaube, man sollte allerdings gar nicht so sehr auf außergewöhnlich hohe Genauigkeit dieser Geräte bedacht sein. Die wesentlichere Aufgabe dürfte die sein, die Orchester dazu zu bringen, diese Geräte auch wirklich zu benützen. Die Lösung dieser Aufgabe würde sehr unterstützt, wenn von maßgebender internationaler Seite aus eine kleine Denkschrift über die Notwendigkeit und den Nutzen der Einhaltung des Normstimmtones ausgearbeitet würde.

Normstimmtonmessungen an etwa 200 zur Prüfung vorgelegten oder zur Kontrolle den Betrieben entnommenen Akkordeons aus den Jahren 1952 und 1953 zeigen, daß von einem Einhalten des Normstimmtones 1952 vielfach nicht die Rede sein kann. Unterschiede von mehreren Hertz innerhalb der Instrumente eines Betriebes kamen öfters vor. Anderseits gibt es Betriebe, die zeigen, daß eine gute Einhaltung des Normstimmtones durchaus möglich ist. Forderungen nach Einhaltung einer Fehlergrenze von $\pm$ 0,8 Hz, die von uns erhoben wurden, erschienen mithin von Seiten des Standes der Industrie und der Eigenschaften der Instrumente (Tab. I) begründet. Die Messungen im 1. Quartal 1953 und darauf folgende Kontrollmessungen zeigen

eine im allgemeinen gute Einhaltung dieser Grenze von $\pm$ 0,8 Hz (= $\pm$ 2 Promille) als ersten praktischen Erfolg. Ähnliche Fehlergrenzen wurden auf Grund zahlreicher Messungen auch für andere Musikinstrumente festgelegt oder sind z.Zt. in Arbeit, mit dem Ziele, vorerst gröbere Fehler zu beseitigen. Es wäre zu erwägen, solche Toleranzen für den Normstimmton nach Schaffung genügender Grundlagen für die wichtigsten Musikinstrumente auch international festzulegen.

2. Notwendigkeit genügender Stimmung von Ton zu Ton. Elektro-optisches Stimmgerät

Die Einhaltung des Normstimmtones allein genügt natürlich nicht. Mindestens ebensoviel Wert ist auf die Einhaltung einer befriedigenden Stimmung der Musikinstrumente von Ton zu Ton zu legen. Diese Stimmung wird seit über 200 Jahren aus guten Gründen im allgemeinen gleichschwebend temperiert gehalten und ist am Musikinstrument wesentlich schwieriger zu verwirklichen als die dem menschlichen Ohr mehr entsprechende reine Stimmung. Temperiert stimmen heißt, mit Ausnahme der Oktave, von Intervall zu Intervall in verschiedenem Maße gegen das eigene natürliche Empfinden zu stimmen. Daß sich hieraus für die Stimmer viele Schwierigkeiten ergeben müssen, leuchtet ein.

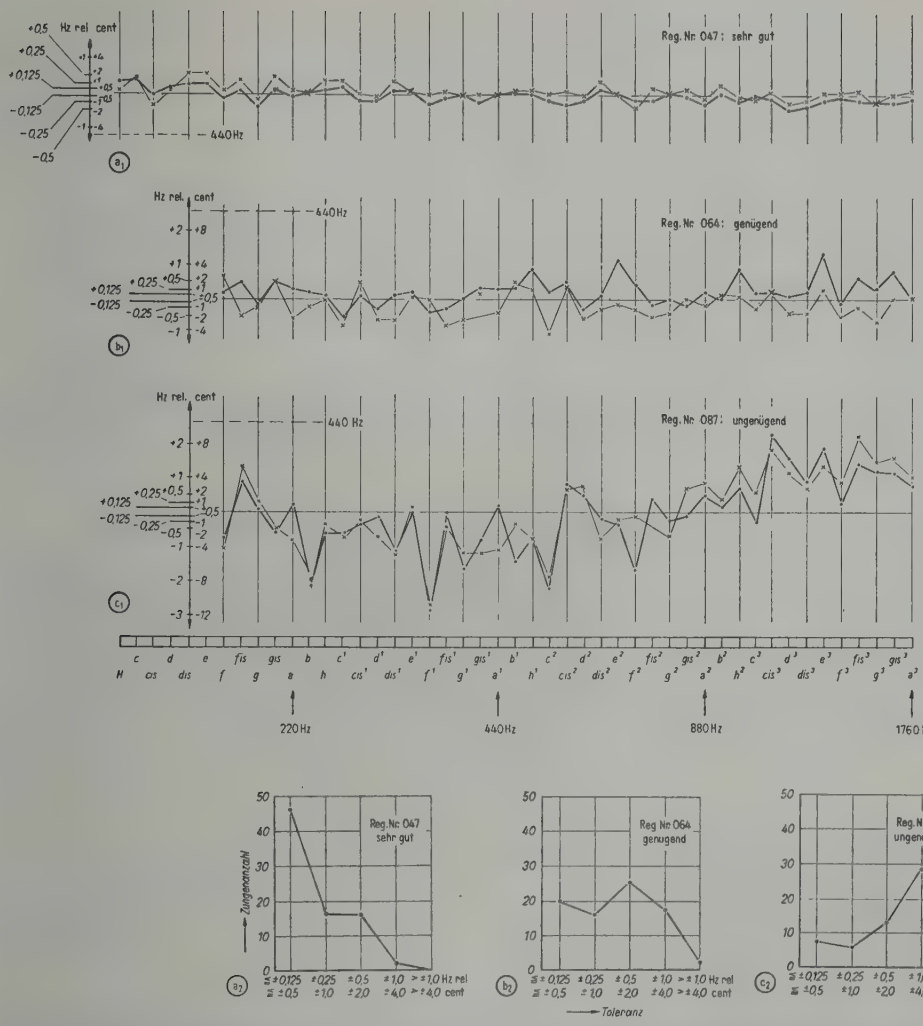
Es lag uns deshalb daran, ein vor allen Dingen praktisch gut brauchbares Stimmgerät zu haben, das vom Ohr unabhängig macht und das sich gleichzeitig für Meßzwecke eignet. Sehr gute Stimmgeräte wurden uns bereits von den Herren Geluk und Young vorgeführt. Wir bauten ein elektronisches Gerät mit einem auch sonst benützten brückenstabilisierten *LC*-Generator [7], der einen Kippgenerator synchronisiert, dessen Spannung an der einen Ablenkplatte einer Kathodenstrahlröhre liegt und auf deren Bildschirm einen Strich in der Frequenz des brückenstabilisierten Tongenerators erzeugt. Dieser Strich wird durch Anlegen einer negativen Spannung an den Wehneltzylinder der Kathodenstrahlröhre gerade wieder zum Verschwinden gebracht. Der von einem Tauchspulenmikrophon aufgenommene und zu messende bzw. zu stimmende Klang geht über einen Verstärker mit Amplitudenbegrenzung an die Kathode der Braunschen Röhre. Die positiven Spannungsimpulse des Klanges erhellen den verdunkelten Strich des synchronisierten Kippgenerators. Es entstehen auf dem Bildschirm Frequenzmarken, die mit der Frequenz zunehmen. Sie laufen nach oben, wenn der zu stimmende Klang zu hoch ist, nach unten, wenn er zu tief ist und bleiben stehen, wenn er richtig ist. So ist das Gerät für den Instrumentenbauer bequem und wirtschaftlich. Die Meßgenauigkeit beträgt $\pm$ 0,3 Promille. Es hat gewisse Vorteile, wie im vorliegenden Falle, sehen zu können, ob ein Ton zu hoch oder zu tief ist. Beim Schwebungshören ist meistens nicht festzustellen, ob der zu stimmende Ton zu hoch oder zu tief ist; der Stimmer muß probieren. Das kostet Zeit und vermindert die Haltbarkeit der Stimmung. Gleichwohl können auch die Sollfrequenzen durch das Tauchspulenmikrophon zum Tönen gebracht werden, doch ist mehr daran gedacht, diese Möglichkeit zusammen mit einem gleichfalls eingebauten Normstimmtongeber dazu zu benützen, genau gestimmte temperierte Intervalle zu Gehör zu bringen und dadurch das Gerät auch für Schulungszwecke verwendbar zu machen.

3. Meßergebnisse

Abb. 1 und 2 zeigen beispielsweise einige Meßergebnisse, auf die hier leider nur teilweise eingegangen werden kann. Waagerecht sind die gemessenen Klänge, senkrecht die Abweichungen von der Sollfrequenz, die in die Abszisse gelegt ist, in Hertz relativ (bezogen auf a^1) aufgetragen. Bei einem vernachlässigbaren Fehler ist dabei 1 Hz rel. = 4 Cent.

Die sehr verschiedene Qualität der Akkordeonstimmung ist offenkundig. Eine nur für Akkordeons geltende Bewertung nach musikalischen Gesichtspunkten gab unser Sachverständigengremium mit „sehr gut", „genügend" und „ungenügend". Damit ist uns künftig eine musikalische Bewertung der Stimmung allein auf meßtechnischer Grundlage möglich. Für andere Musik-, insbesondere Blasinstrumente, gelten andere musikalische Maßstäbe; im allgemeinen sind die Anforderungen niedriger. Ordnet man noch die vom Stimmer gemachten Fehler in verschiedene Bereiche ein ($\pm \frac{1}{2}$ Cent, $\pm$ 1 Cent usw., Abb. 1), so zeigt sich das erstaunliche Ergebnis, daß unsere besten Akkordeonstimmer sogar Fehler beim temperierten Stimmen hören, die $\leq \pm \frac{1}{2}$ Cent sind, denn die Mehrzahl der Fehler liegt innerhalb dieses Bereiches. Beste Klaviere und Flügel (Abb. 2) werden nach unseren bisherigen Ergebnissen in den zwei mittleren Oktaven sogar noch etwas genauer gestimmt, als die entsprechende Fehlerbereichskurve zeigt. Sie zeigen meistens eine aufsteigende Tendenz [8] ihrer Stimmung, die zur größeren Brillanz des Klanges führt. Die von Herrn Young gezeigten Kurven weisen eine gleiche Tendenz auf.

Die bisher gemessenen Orgelstimmungen reichen erklärlicherweise nicht an die eben gezeigten besten Stimmergebnisse heran. Blasinstrumente wären im Vergleich mit Akkordeons und Flügeln sogar als ausgesprochen schlecht gestimmt zu bezeichnen. Gute Spieler überwinden zwar in langsam zu spielenden Musikstücken die Stimmungsfehler ihrer Instrumente weitgehend, würden aber auf einwandfrei gestimmten Instrumenten besser spielen können.



Toleranz Hz rel.	Bewertung	Anzahl der Zungen		Punkt-		Anzahl der Zungen		Punkt-		Anzahl der Zungen		Punkt-	
		gepr.	red.	wertung	zahl	gepr.	red.	wertung	zahl	gepr.	red.	wertung	zahl
$\leq \pm 0{,}125$	sehr gut	59	46	— 0	— 0	20	20	— 0	— 0	8	8	— 0	— 0
$\leq \pm 0{,}25$	gut	22	17	— 1	—17	16	16	— 1	— 16	6	6	— 1	— 6
$\leq \pm 0{,}50$	mittel	23	17	— 3	—51	26	26	— 3	— 78	13	13	— 3	— 39
$\leq \pm 1{,}00$	schlecht	2	2	— 8	—16	18	18	— 8	—144	29	29	— 8	—232
$> \pm 1{,}00$	sehr schlecht	0	0	—20	— 0	2	2	—20	— 40	26	26	—20	—520
					—84				—278				—797

Abb. 1. Akkordeonstimmungen verschiedener Güte.

a_1, b_1, c_1: Abweichungen von der Sollfrequenz, bezogen auf den arithmetischen Mittelwert;
·—·: Zug, x—x Druck.

a_2, b_2, c_2: Häufigkeit (geprüfte bzw. reduzierte Anzahl) der Zungen in den verschiedenen Toleranzen.

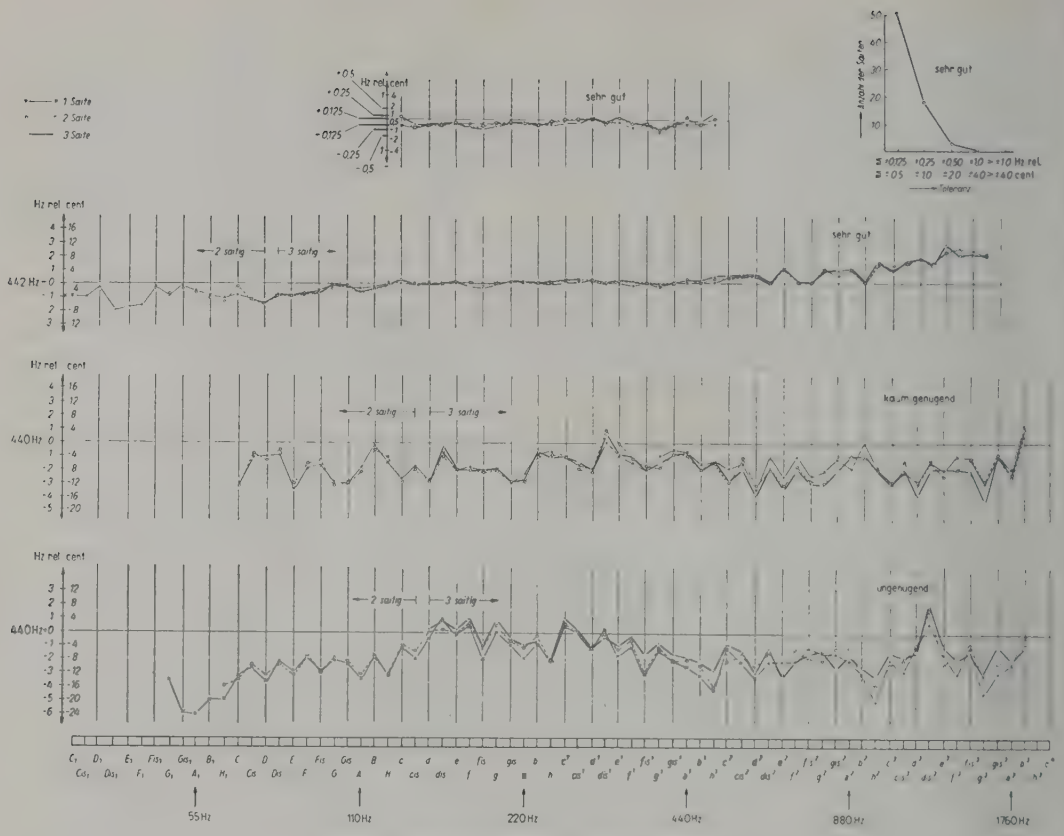


Abb. 2. Klavierstimmungen verschiedener Güte.

Die Kurven zur Stimmung gewinnen damit beachtliche Bedeutung für den Instrumentenbau.

Es soll schließlich nicht versäumt werden, zu erwähnen, daß unsere Meßergebnisse zur Akkordeon- und Klavierstimmung im gewissen Sinne in Zusammenhang stehen mit früheren Ergebnissen [9], [10], [11] zur Unterschiedsschwelle der Tonhöhe. Diese verläuft im Bereich von etwa 600 ··· 3000 Hz linear und steigt nach tieferen Frequenzen stark an. In unserem Falle reicht dagegen bei Akkordeons der lineare Teil noch bis zu etwa 90 Hz. Außerdem beträgt die Unterschiedsschwelle der Tonhöhe im linearen Teil etwa 3 Promille [10], [11], während sich Fehler der temperierten(!) Stimmung von $\pm$ 0,3 Promille oder noch kleiner hören lassen.

Wenn man annimmt, daß diese früheren Ergebnisse im wesentlichen auch heute noch gelten, würden sie zum Mechanismus des Stimmens zeigen, daß sich das Stimmen der tiefen und mittleren Klänge hauptsächlich auf die hohen Teiltöne stützt. Zur größeren Hörgenauigkeit beim temperierten Stimmen haben sich jedoch bisher keine Gesichtspunkte finden lassen, die diese erklären konnten, so daß es vielleicht doch zweckmäßig wäre, die Messungen zur Unterschiedsschwelle der Tonhöhe mit guten Berufsstimmern zu wiederholen. E. Meyer (l.c.) weist in seinem Handbuchaufsatz nicht zu Unrecht darauf hin, daß in diesen Fragen Übung und musikalische Veranlagung eine ganz wesentliche Rolle spielen.

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THE EFFECT OF VOICING ADJUSTMENTS ON THE TONE QUALITY OF ORGAN FLUE PIPES

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Summary. Details are given of the tonal effects of the chief adjustments which may be made to a pipe. The effect of pipe diameter in limiting the number of harmonics is explained. Edge tones are shown to be confined to the initial sounds emitted by a pipe, and a new theory of operation of the flue pipe is put forward.

Sommaire. On indique en détail l'influence, sur la qualité du son, des principaux réglages qu'on peut faire sur un tuyau d'orgue. On explique comment le choix du diamètre du tuyau permet de limiter le nombre d'harmoniques. On montre que les sons d'arête n'apparaissent qu'au début de l'émission sonore d'un tuyau, et on propose une nouvelle théorie du fonctionnement du tuyau à embouchure.

Zusammenfassung. Es werden Einzelheiten über die klanglichen Auswirkungen der Bemessung und Einrichtung von Orgelpfeifen angegeben. Die Begrenzung der Zahl der Obertöne durch richtige Wahl des Pfeifendurchmessers wird erklärt. Es wird gezeigt, daß Schneidentöne auf die Einschwingvorgänge der Orgel beschränkt sind, und eine neue Theorie der Arbeitsweise der Lippenpfeife wird entwickelt.

1. Introduction

This investigation was confined to open metal cylindrical pipes ("open diapason") 0.05 m diameter, 0.64 m long, giving 261 c/s. Measurements were made in a small anechoic room. Steady tones were analysed with a Marconi heterodyne wave-analyser and a Brüel & Kjaer level recorder, and starting transients with a single stroke C.R.O. The microphone (crystal) was 0.05 m from the mouth, thus measuring the sound pressure due only to the mouth.

2. Voicing adjustments

Referring to Fig. 1, air under pressure passes through the foothole and escapes from the flue, interacting with the upper lip. "Voicing" operations are needed to make the pipe sound its note properly [1]; the principal ones examined were as follows:

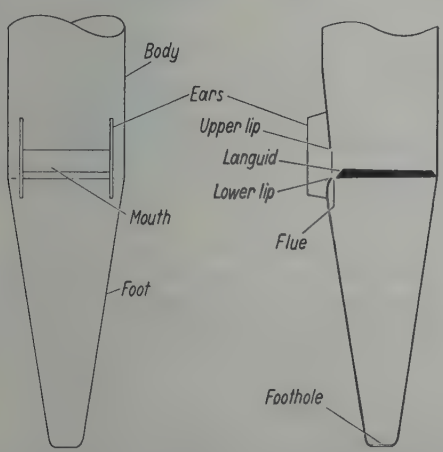


Fig. 1. Diagram of lower part of typical flue pipe.

a) "Nicking", i.e. cutting notches in the edges of the flue, is found chiefly to reduce the background noise (Fig. 2). The explanation is suggested that the sharp velocity gradient at the edge of the windstream causes turbulence and small eddies enter the pipe irregularly, giving random noise. The air flowing through the nicks reduces this velocity gradient, and reduces the turbulence. Pipes with low windstream velocities (Reynolds' number $< 1\,000$) do not need nicking.

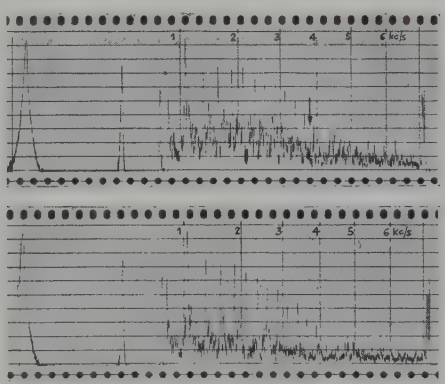


Fig. 2. Top, un-nicked pipe; bottom, nicked pipe. The arrow in top figure indicates the theoretical "limiting frequency".
There is an increase in sensitivity of 30 dB after the 3rd harmonic.

b) Increasing the foothole increases the number and intensity of the harmonics, while

c) "Cutting up" the mouth, to make the upper lip higher, reduces them. Neither of these alterations, however, greatly affects the fundamental.

d) Altering the height of the languid edge in relation to the upper lip has a great influence on the

tone. If it is high, the harmonics are increased and the initial transient (which is important musically) is built up slowly (Fig. 3), and vice versa.

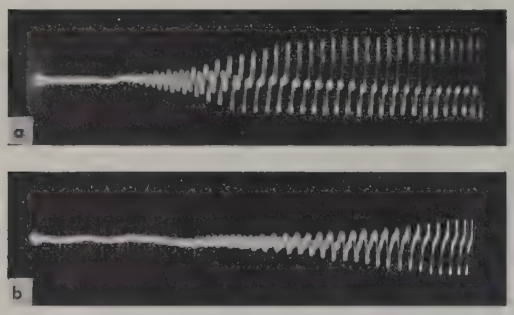


Fig. 3. Starting transients: a) low languid; b) high languid.

3. Effect of pipe diameter

The chief damping in a flue pipe is by radiation, both at the open end and at the mouth. Considering each as a piston radiating in a baffle, there is a phase change at each [2] dependent on frequency, and at certain frequencies the total phase change of a wave reflected first at the open end and then at the mouth will be 180°. Such frequencies cannot develop. The theoretical "limiting frequency" is marked in Fig. 2. A larger pipe diameter or a larger mouth area gives a greater phase change, and thus a lower "limiting frequency".

4. Edge-tones

If the body is cut off a pipe, leaving only the upper lip, a complex edge-tone is produced. The windstream velocities at which any of its components coincide with any of the first few harmonics of the pipe are found to be those at which the pipe "whistles" before it comes to full speech. At the windstream velocities for normal speech, however, the edge-tone frequencies are far above the fundamental, and appear to have little influence on the higher harmonics. It thus appears that edge-tones play little part in normal speech.

5. Operation of flue pipe

A new theory will now be postulated, being similar to the older "air-reed" theories, but representing a considerable advance on them.

The windstream from the flue induces an airflow from its sides, behaving qualitatively as a "Bickley jet" [3], Fig. 4b. Measured velocity profiles are shown in Fig. 4a. It thus abstracts air through the mouth from inside the pipe. This causes a rarefaction, and the windstream, which is normally directed outwards, is forced to blow into the pipe. This causes a compression and it is forced to blow out again, to continue the cycle.

If it be assumed that the velocity at the edge of the upper lip (a-b in Fig. 4a) represents the initial rate of abstraction in each cycle, the natural frequency of this process can be approximately calculated, and agrees reasonably well with observation. Normally, however, the standing-wave system in the pipe will exert considerable stabilisation; but if, due to a higher windstream velocity and hence more rapid abstraction, the "mouth-frequency" is too high, the pipe may ultimately sound a harmonic.

This condition may be cured by pushing in the upper lip or cutting it up more; in each case, due to the shapes of the velocity profiles, the velocity at the edge of the upper lip will be lower, giving slower abstraction. An obstruction placed outside the mouth gives a similar effect, since it impedes the induced airflow and thus causes the windstream to be directed further outwards.

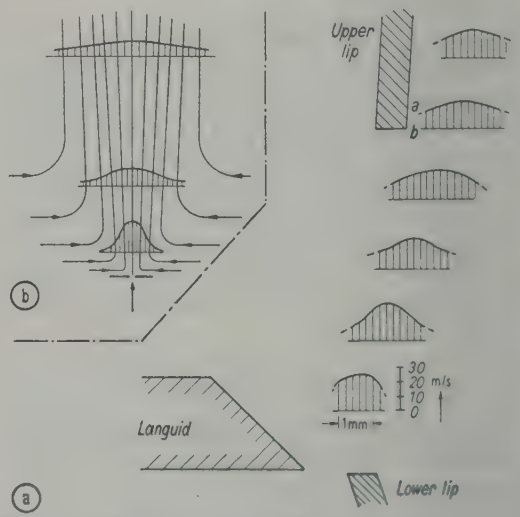


Fig. 4. (a) Measured velocity profiles in windstream, (b) Theoretical two-dimensional jet.

The sound intensity is approximately calculable, since the peak rarefaction must deflect the jet sufficiently to blow inside the pipe. Such calculations agree fairly well with observation. Clearly intensity will increase with windstream velocity.

Further details of this work are being communicated elsewhere.

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[1] Mercer, D. M. A., J. acoust. Soc. Amer. 23 [1951], 45.
[2] Morse, P. M., Vibration and Sound, 2nd ed. McGraw Hill, New York 1948, p. 255.
[3] Bickley, W. G., Phil. Mag. 23 [1937], 727.

Discussion

Gavreau, V.: Nous étudions actuellement le fonctionnement de sifflets qui peuvent être considérés comme le cas limite

d'un tuyau d'orgue et j'ai trouvé expérimentalement en décomposant un sifflet réel en ses parties constitutives: "jet"-"couteau"-"résonateur" qu'aucun son n'était émis tant que ces éléments étaient séparés, ce qui démontre l'impossibilité d'expliquer le fonctionnement d'un sifflet de police ordinaire par la théorie du "son du jet" ou "son de couteau". On est ainsi obligé d'admettre l'exactitude de la théorie classique de HELMHOLTZ et de Lord RAYLEIGH. La théorie des tourbillons et des sons de couteau n'est valable que pour les vitesses d'air dépassant largement celles qu'on utilise dans les sifflets et dans les instruments de musique.

THIENHAUS, E.: Wie groß ist der Winddruck bei den untersuchten Pfeifen?

LECTURER: Ungefähr 85 mm Wassersäule unter dem Pfeifenfuß, entsprechend ca. 45 mm ($= h$) unter dem Kern. Die Windgeschwindigkeit ergibt sich aus: $v \simeq 4\sqrt{h}$ (v in m/s, h in mm).

WELCHE MÖGLICHKEITEN BESTEHEN FÜR EINE SINNVOLLE ANWENDUNG ELEKTRONISCHER MUSIKINSTRUMENTE?

von W. MEYER-EPPLER

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Zusammenfassung. Die elektronischen Musikinstrumente geben zusammen mit den Tonspeicherungsverfahren und den Schaltelementen der Tonfrequenztechnik dem Komponisten ein neues Gestaltungsmittel in die Hand. Aufgabe des Akustikers ist es, den Komponisten über die sich darbietenden Möglichkeiten zu unterrichten. Ein beim Nordwestdeutschen Rundfunk in Köln eingerichtetes „elektronisches Studio" soll der Verwirklichung neuer Kompositionsgedanken dienen.

Summary. Electronic music instruments give by the choice of tone components a new medium to the composer. It is the business of the acoustic expert to educate composers in these possibilities. The Cologne Broadcasting Station has built a special "electronic studio" for the use of composers.

Sommaire. Les instruments de musique électroniques offrent aux compositeurs un nouveau moyen d'expression, grâce aux procédés d'enregistrement et de la technique des audio-fréquences. Il incombe à l'acousticien de montrer aux compositeurs les possibilités qui leur sont offertes dans ce sens. Un "Studio électronique" installé à Cologne, dans les Services de la Radiodiffusion du Nord-Ouest de l'Allemagne, doit précisément servir à étudier de nouveaux modes de composition musicale.

Obwohl die elektronischen Klangerzeuger so alt sind wie die Verstärkertechnik und der Rundfunk, und obwohl sie verschiedentlich Lösungsformen gefunden haben, die dem Naturwissenschaftler oder Techniker reizvoll erscheinen können, ist ihre Wirkung auf die Musik bisher praktisch ohne Bedeutung geblieben. Fragt man nach den Ursachen für die Ablehnung der von der Industrie und von Erfindern angebotenen Klangerzeuger durch die Musiker, so erfährt man in der Regel, es gebe bereits so hochvollendete mechanische Musikinstrumente, daß jeder Versuch, diesen etwas Gleichwertiges auf elektrischer Grundlage an die Seite zu stellen, zum Scheitern verurteilt sein müsse. Diesem Argument könnte man beipflichten, wenn es wirklich der Zweck eines elektronischen Musikinstruments wäre, Klänge zu erzeugen, die den herkömmlichen Instrumentalklängen ähneln oder diese gar imitieren. Seltsamerweise nun haben viele Erbauer von elektronischen Klangerzeugern den Ehrgeiz, gerade solche Instrumentalimitatoren zu schaffen. So wenig künstlerischen Sinn es jedoch hätte, Pseudo-Bronzeplastiken aus passend gefärbtem Polyvinylchlorid herzustellen, so absurd ist der Gedanke, das Violinkonzert von Beethoven auf einer elektronischen Pseudo-Violine zu spielen. Jedem Werkstoff und jedem Instrument kommt eine ihm allein angemessene Bearbeitungs- oder Behandlungsweise zu, und die bloße Übernahme einer bewährten künstlerischen Gestaltungstechnik in ein ganz anders geartetes Gebiet genügt nur selten den berechtigten Ansprüchen auf eine *materialgerechte* künstlerische Form.

Die für traditionelle Instrumente geschriebene Musik muß mithin gegen jeden Versuch geschützt werden, ihr elektronische Instrumental-Imitatoren aufzuzwingen; eine Ausnahme bildet möglicherweise die Unterhaltungsmusik. Ganz anders aber ist die Lage, wenn einer der *lebenden Komponisten* sich entschließt, ein ihm geeignet erscheinendes elektronisches Instrument bei der musikalischen Gestaltung zu verwenden. Welchen Gebrauch er von diesem Instrument macht, ist eine Angelegenheit, die er allein zu vertreten hat. Vom naturwissenschaftlichen Forscher und insbesondere dem Akustiker erwartet er nicht Billigung oder Ablehnung seines Vorhabens, sondern fachliche Unterrichtung. Ob der spätere Hörer dann das Ergebnis der Beschäftigung mit den vom Akustiker bereitgestellten Kompositionsmitteln als Musik oder Geräusch oder sonstwie bezeichnet, liegt außerhalb des *wissenschaftlichen* Diskussionsbereichs.

Es soll im folgenden versucht werden, einige der Kompositionsmittel zu nennen, die für die elektronischen Klangerzeuger kennzeichnend sind und die kein Gegenstück bei den klassischen Musikinstrumenten und Kompositionsmethoden haben [1].

Von den mechanischen unterscheiden sich die elektronischen Instrumente hauptsächlich dadurch, daß sie nicht *Schall*erzeuger, sondern *Strom-* oder *Spannungs*erzeuger sind. Die Übersetzung in den Schallbereich muß durch elektroakustische Wandler, z.B. Lautsprecher, vermittelt werden. Da nun die elektrischen Ströme die zu verarbeitende Materie darstellen, liegt es nahe, als Kompositionsmittel die in der Tonfrequenztechnik gebräuchlichen Erzeugungs-, Mischungs- und Umwandlungsverfahren einzusetzen. Als besonders naheliegend muß man also die Gestaltung im *elektrischen* Bereich ansehen, und es wäre ein — vielleicht gelegentlich einmal zu rechtfertigender — Umweg, wollte man die elektrischen Schwingungen zuvor in Schall verwandeln.

Einige der bereits bei Kompositionsversuchen verwendeten elektrischen Gestaltungsmöglichkeiten seien hier angeführt. Die hochentwickelten Speicherungsverfahren (Lichtton, Nadelton, Magnetton) bieten die Möglichkeit, den Zeitmaßstab bei der Herstellung der Komposition anders zu wählen als bei der Wiedergabe. Im Grenzfall kann die Bearbeitung der Komposition bei *ruhendem* Schallträger erfolgen (z.B. durch Zeichnen der Tonspur mit Pinsel und Feder oder durch Aneinanderkleben von Bandstücken); der musikalische Kompositionsprozeß gewinnt dann Ähnlichkeit mit der Maltechnik. Der Geschwindigkeit in der Aufeinanderfolge von Schallereignissen sind bei diesem Verfahren keine Grenzen gesetzt, und so nimmt das Ergebnis leicht den Charakter einer äußerst virtuosen Aufführung an, obgleich von einer „Aufführung" im üblichen Sinne natürlich keine Rede sein kann; das vom Komponisten in allen Einzelheiten ausgearbeitete und in verbindlicher Form — etwa als Magnettonband — vorgelegte Werk bedarf ja keiner Interpretation durch einen ausübenden Künstler mehr.

Eine solche „authentische" Bandkomposition ist der Wiedergabe durch den Rundfunk artgemäßer als etwa die bisher übliche Musikübertragung aus dem Konzertsaal [2], ohne daß damit der letzteren die Berechtigung als Unterrichtungs- und Belehrungsmittel abgesprochen würde.

R. Bierl [3] hat darauf hingewiesen, daß viele der als „Musikinstrumente" angebotenen elektronischen Klangerzeuger grundsätzlich nichts zu leisten vermögen, was nicht auch mit mechanischen Instrumenten zu erreichen wäre. Dies gilt jedoch für die mit Speicherverfahren arbeitende Kompositionstechnik nicht mehr. Auch die Einführung nichtlinearer Elemente (Gleichrichter, Modulatoren, Elektronenschalter) in das Kompositions-Instrumentarium stellt eine im mechanischen Bereich nur schwer nachzubildende Maßnahme dar. Die am stärksten von allen gewohnten Klängen abweichenden Wirkungen erzielt man demgemäß gerade mit solchen *nur im elektrischen Bereich* wirksamen Mitteln.

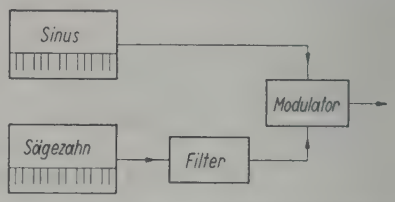


Abb. 1. Anordnung zur multiplikativen Mischung.

So lassen sich z.B. die von zwei Schwingungsgeneratoren erzeugten Schwingungen dadurch in ungewohnte Beziehungen zueinander bringen, daß man sie nach Abb. 1 miteinander *multipliziert*. Bei ganzzahligem Frequenzverhältnis der beiden Komponenten ist das Ergebnis eine periodische Schwingung von eigenartiger spektraler Zusammensetzung; man kann es beispielsweise so einrichten, daß alle durch 3 oder 4 oder 5 teilbaren Harmonischen wegfallen. Eine geeignete Nomenklatur für die so entstehenden Klänge fehlt naturgemäß noch. Ersetzt man eine der periodischen Schwingungen durch farbiges Rauschen, so ergeben sich akustische Wirkungen, die zwischen reinem Klang und reinem Geräusch liegen und der einen oder anderen Grenze beliebig genähert werden können.

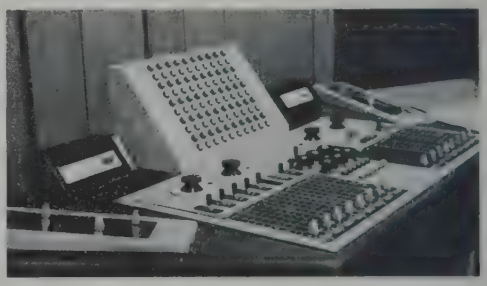


Abb. 2. Steuerpult im elektronischen Studio des NWDR Köln.

Ein *elektronisches Studio*, das dem Komponisten die selbständige Gestaltung von Bandkompositionen ermöglicht, kann naturgemäß nur wenig Ähnlichkeit mit einem Studio haben, in dem in herkömmlicher Weise musiziert wird. Abb. 2 und 3 zeigen Ausschnitte aus dem von F. Enkel beim Nordwestdeutschen Rundfunk in Köln eingerichteten Studio; es ist mit Schwingungsgeneratoren nach F. Trautwein („Trautonium") und H. Bode („Melochord")



Abb. 3. Bode-Melochord im elektronischen Studio.
a = Aussteuerungsmesser, b = Klangfarbenmanual,
c = Tonhöhenmanual (schwarze Tasten links von b und c dienen zur Oktavenwahl).

von einer den besonderen Anforderungen der Bandkomposition entsprechenden Konstruktion ausgerüstet. Eine automatische, von F. Enkel entwickelte Schaltanlage gestattet es dem im Studio tätigen Komponisten bzw. dem ihn unterstützenden Tontechniker, die jeweils erforderlichen Schaltungen (Verbindung der Klanggeneratoren untereinander und mit den verschiedenen Aufnahme- und Wiedergabemaschinen, Filtern, Verhallern, Verzerrern usw.) rasch und sicher auszuführen.

Schrifttum

[1] Meyer-Eppler, W., Melos 20 [1953], 5.
[2] Meyer-Eppler, W., Techn. Hausmitt. NWDR 4 [1952], 130.
[3] Bierl, R., Phys. Verh. 3 [1952], 175.

Diskussion

Winckel, F.: In der oberen Stimme des vorgeführten Bandes ist ein Tremolo zu hören, in der unteren Stimme ein ziemlich stationärer Vorgang. Worauf ist dies zurückzuführen?

Vortragender: Das Tremolo kann willkürlich in jede Stimme eingeführt werden. Es sollte hier die Unterscheidung der beiden Stimmen erleichtern.

THE CHARACTERIZATION OF SOUND OBJECTS BY USE OF THE LEVEL RECORDER IN MUSICAL ACOUSTICS

by A. Moles
Centre d'Études Radiophoniques, Paris

Summary. A musical sound is a quasi-periodic phenomenon, best expressed by a tri-dimensional representation in terms of level, pitch, duration, the perception of which follows Fechner's law. The logarithmic bathymeter only gives recordings of the level-time plane, under various restrictions, so we have developed a method of classification of the signals on punched cards which can be applied either to noises or musical sounds.

Sommaire. Le signal musical est un phénomène non périodique à degré de périodicité élevé, exprimé le plus adéquatement par la représentation tridimensionnelle: Niveaux, hauteurs, temps ou durées, la perception de ces trois grandeurs suivant la loi de Fechner. Le bathymètre logarithmique enregistre seulement le plan Niveaux-Durées avec diverses restrictions de principe, exprimées théoriquement par la notion d'"Intégrale de lissage" logarithmique:

$$\overline{f(z)} = \frac{1}{2\varepsilon z} \int_{(1-\varepsilon)z}^{(1+\varepsilon)z} f(z)\, \mathrm{d}z.$$

Pour les applications à l'étude du signal musical, nous avons développé une méthode générale de classification des objets sonores sur cartes perforées qui permet une sélection ultérieure à partir de tel ou tel critère désiré a priori. Une telle méthode peut être utilisée pour des études quantitatives des bruits et des sons musicaux.

Zusammenfassung. Musikalische Schallvorgänge stellt man am besten dreidimensional durch Pegel, Tonhöhe und Zeitdauer dar, da die Empfindungen dieser drei Größen dem Fechnerschen Gesetz gehorchen. Dazu läßt sich ein logarithmischer Pegelschreiber verwenden, der nur die Ebene Pegel-Zeit aufzeichnet. Weiter wurde eine allgemeine Klassifikationsmethode unter Verwendung von Lochkarten entwickelt, die eine spätere Auswahl nach bestimmten Gesichtspunkten ermöglicht und bei quantitativen Untersuchungen an Geräuschen und musikalischen Vorgängen verwendet werden kann.

Let us start with the assumption that musical signals are not strictly periodic. In fact, they are a sequence of transients with a high degree of periodicity (this is the mathematical expectancy of knowing what will follow after what we know *now* of the phenomenon). This idea, on statistical grounds, can be expressed by the auto-correlation integral of Wiener.

The perception of the instantaneous degree of periodicity is a psychophysiological process in the

upper centres of the brain — a process not clearly explained hitherto. We will retain as fundamental rule that an objective study *must* include, as R. VERMEULEN clearly said in his general lecture, a psychological approach.

A musical signal as a particular kind of "sound stuff" requires first an adequate description. It is often impossible with very complex sources such as the orchestra — especially in most recent music (STRAVINSKY, SCHÖNBERG, CHAVEZ) to define clearly the pitch of the signal, and the concept itself of pitch, although clear as regards separate instruments, becomes indistinct.

There remains only a "colour" which is the direct perception of the spectrum shape, and the evolution of spectra in time is the chief objectively unequivocal concept.

The most adequate representation is the tridimensional: level, pitch, time (Fig. 1), which we have discussed in previous publications. We will naturally use for the axes physical quantities closely related to FECHNER's law — taken as the first approximation to psycho-perception.

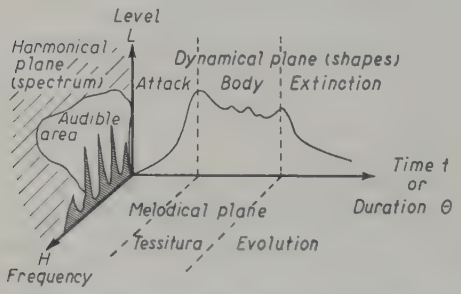


Fig. 1. Three-dimensional diagram of the complex tone.

$$L = 20 \log \frac{A}{A_0} \text{ dB}, \qquad H = \frac{1}{0.301} \log \frac{f}{f_0} \text{ octaves.}$$

A = sound pressure,
A_0 = threshold of sound pressure feeling ($2 \cdot 10^{-5}$ N/m^2),
f = frequency in c/s,
f_0 = threshold of frequency feeling (16 c/s approximately).

As regards the scale of the time axis either we are interested in the signal *as a whole* for instance in a record disk, or a magnetogram, when there is no a priori reason for giving special interest to any particular instant: accordingly, we shall divide the time axis with seconds marks, corresponding to the length of a magnetogram; or, alternatively, we are dealing with the properties of one of the transient phenomena whose sequence builds up the signal and we study it in relation to its influence on the individual and his perception. In that case, we will consequently be interested not in the time expansion, but in the evolutive perception of this transient or group of transients, involving the notion of *duration* as fundamental element. We will be led therefore to apply general sensation rules, especially the FECHNER law — disregarding its inaccuracy — and to use a logarithmic scale:

$$\theta = K \log \frac{t}{t_0}$$

t_0 being what psychologists call the "thickness of the present", i.e. the short duration where two distinct physical phenomena appear simultaneously to the mind, in other words the threshold of time perception (1/10 ··· 1/20 s).

Such a tridimensional diagram, in which we point out the homogeneity of expression of the three scales, will express adequately the relative importance of different parts of a whole, which we will call *sound object* or *sound cell* according to its complexity, bringing out more correctly the relative importance of the initial transients or consonants in speech, which is here considered as a peculiar kind of musical signal.

Naturally, such a division is not arbitrary, it may be given an objective basis. The best way of testing this idea is to cut off random lengths from a magnetogram and play them separately. It is then noticed that some definite zones of length bring more interesting results than others as representing the auditory sensation of a whole. Among them are the sentences in speech (definite duration of about 5 to 10 seconds), which are complex organized forms which we shall call *cells* and other shorter snippets, 1 second approximately, which possess a centre of interest and define what we have called *sound objects* in which the spectrum follows a continuous evolution after a *type*. Naturally, these divisions will only occasionally coincide with the tones and notes of scores: the complex tone is, not infrequently, given as a whole in the duration of 0.7 s, corresponding to a loop on a 78 rev/min record.

We have been led to make a critical study of the properties of the tridimensional diagram, especially concerning transient shapes.

We know that the galvanometer system in a high-speed level recorder is able to write correctly up to 1 000 dB/s $\Delta L/\Delta t$ phenomena, but it must be pointed out that the system acting on the galvanometer coil is a rectifier and consequently integrates after its own personal point of view, the complex phenomena which can be applied to the input terminals. Consequently, if we have been able to verify with a cathode-ray oscillograph the performance quoted above for sine waves with simple laws of increment, we have noticed many differences in the details of the curves of very fast increment such as transients of wind or string instruments: the average $\Delta L/\Delta t$

being approximately the same as on the oscillograph film records, but the shape being, above 400 dB/s, widely different from

$$\frac{\mathrm{d}\log A^2}{\mathrm{d}t} = \frac{\mathrm{d}L}{\mathrm{d}t}.$$

This fact is especially noticeable when there is a sudden change in the spectral composition of the transient, as is the case with pizzicati or staccati of strings, in which a line spectrum merges out of the initial continuous spectrum. This would theoretically lead to the conclusion that the bathymeter, even the high speed model, is an inadequate instrument for transient studies and that we are thrown back on elaborate logarithmic transformations of oscillograph films — a typically awkward method, not to speak of the price of films — especially considering the complicated shapes of transients given by attacks of musical instruments which are, as BACKHAUS has shown a long time ago, appreciated by the mind as a whole, as what psychology calls a "Gestalt". But it must be always borne in mind that only the psychophysiological feelings interest us in such studies, and this gives a new approximation tolerance: the one within which the ear does not feel noticeable level variations. We have shown in a previous study that there is at least a tolerance margin of $\pm$ 2.5 dB, which means that any shape given by an apparatus is correct when it differs from the real shape by less than $\pm$ 2.5 dB, and probably much more in transient states, the above figure having been established for slowly varying acoustical signals (Fig. 2).

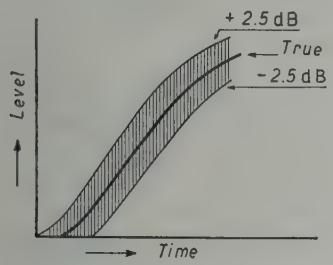


Fig. 2. Margin of tolerance unnoticeable to the ear.

This consideration enlarges the tolerance of the perhaps too strict BACKHAUS point of view and leads to a doctrine as to the possible uses of the high-speed bathymeter for transient studies; for instance it seems quite right to make use of it for a large number of musical instruments and consonant phonemes (Fig. 3). The same phenomenon of "time averaging" naturally occurs in the "body" of the complex tone or phoneme but practically to a lesser extent so that it is generally negligible with a good writing speed adjustment.

A general study of all the previous phenomena of averaging by any type of recording apparatus can be made by the consideration of the two "smoothing functions" as follows:

1) Linear smoothing integral:

$$\overline{f(\omega)} = \frac{1}{2\Delta\omega} \int_{\omega-\Delta\omega}^{\omega+\Delta\omega} f(\omega)\,\mathrm{d}\omega$$

2) Logarithmic smoothing integral:

$$\overline{f_{\log}(\omega)} = \frac{1}{2\varepsilon\omega} \int_{\omega(1-\varepsilon)}^{\omega(1+\varepsilon)} f(\omega)\,\mathrm{d}\omega$$

here written with ω as variable, which is especially the case in frequency-response measurements, but having analogous expressions in t as variable or with any other variable. This notion of smoothing integral applies to *any* system of recording.

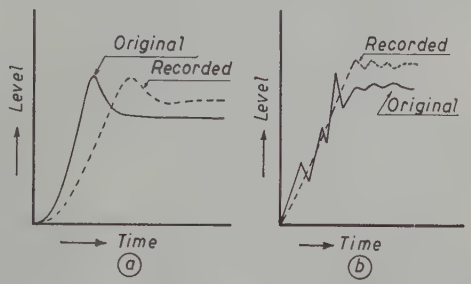


Fig. 3. The kind of errors allowed to the high speed level recorder;
(a) wrong, (b) satisfactory.

We shall conclude by an example of an application to a problem which assumes a growing importance in new electronic and concrete music, viz. the *classification of sound objects*. Music being defined as a time dialectic, we are led to consider it as the assembly of sound shapes in a structural pattern. Analytical study of traditional orchestral music or synthetic experiments on the creation of new musical sequences focuses attention on a large number of sound objects removed from their time sequence.

Such sound shapes must be considered as basic material and consequently classified in a way allowing one to find any of them out of the stock as required. This involves a special problem of classification. We may require for instance "any cell of four notes with sharp transients and no reverberation, decrescendo along the four, of mean tessitura and timbre typical of Mozart's classical orchestra", or for "any steam locomotive isolated tone, with a definite inharmonic line spectrum and with the component lines reverberated, having a fundamental of C".

We have tried to solve this problem by proposing a repertory of typical criteria of the sounds studied and classifying them on a punched-card system, allowing one to select on request any sound object showing such or such criteria. Naturally, it must be

pointed out that among the essential criteria of sound diagrams only a part is relative to the L, t plane, consequently a great many interesting criteria, especially in the colour of the sound, do escape this classification, but the system of punched cards presents the peculiarity of allowing any widening of classification without destroying previous work. Besides, we can in such a way, make the card an *identifying card* perfectly characteristic of the sound object, i. e. giving a biunivocal correspondence between sound shape and card. We quote typical shape characteristics:

General	Initial transient	Body of complex tone	Final transient
Duration	aeolian	vibrated	sudden
Symmetry	abrupt	undulated	smooth
	steep	constant	beating
	gradual	declining	reverberated, steep
	pinched	mosaic	reverberated, gradual
	artificial	evoluting	coloured
	stepped up	clipped	

Each card has a "criteria" capacity of about 120 binary characters (yes, no).

Practically, the classification possibilities may be widely extended and brought closer to the musician's needs by making with the level recorder three different diagrams:

1) bass under 400 c/s,
2) medium between 400 and 1 600 c/s,
3) treble above 1 600 c/s,

and our problem now is to make a closer connexion between the identification cards and the utilization cards. These cards are assembled in two files by means of metal rods, which are placed through the perforated holes expressing the above criteria either in a qualitative or quantitative manner. This makes the "identifying filing box". We must then build a new "utilization filing box" which will provide the criteria specifically interesting the musician (J. POULLIN).

In conclusion, we must point out:

a) Such a classification, here presented only from the qualitative point of view, may be without any difficulty changed to a quantitative one. It suffices to leave enough punched holes for numerical expressions, e. g. of extinction time and body length.

b) Such a classification which has been made for the purposes of concrete music and musical signal studies is naturally valid for any identification of sound objects, especially for: noise studies, functional studies of musical instruments, electronic music, speech identification.

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RÉCENTS RÉSULTATS OBTENUS DANS L'ÉTUDE ÉLECTROACOUSTIQUE DE LA CAISSE HARMONIQUE DES INSTRUMENTS À ARCHET

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Sommaire. Plusieurs chercheurs ont employé des méthodes électro-acoustiques pour l'étude des instruments à archet. En particulier à l'Istituto di Ultracustica "O. M. Corbino" (Rome) on a développé une méthode pour la mesure de l'amplitude de vibration de la caisse harmonique de ces instruments, dont on excite les vibrations forcées au moyen d'une force sinusoidale à amplitude constante et à fréquence variable. L'examen des courbes caractéristiques, amplitude de vibration-fréquence, conduit à des conclusions très importantes à l'égard des qualités musicales des instruments. L'évaluation de ces qualités est réduite à un jugement objectif qui concerne essentiellement les vibrations élastiques de la seule "table d'harmonie". On donne aussi les résultats obtenus dans un grand nombre d'essais sur des violons de construction moderne ou restaurés.

Zusammenfassung. Nachdem schon mehrfach elektroakustische Methoden zur Untersuchung von Streichinstrumenten benutzt wurden, hat man speziell im Istituto di Ultracustica „O. M. Corbino" (Rom) eine Methode zur Messung der Schwingungsamplitude des Klangkörpers solcher Instrumente entwickelt, bei der man mit sinusförmiger Kraft konstanter Amplitude und variabler Frequenz erzwungene Schwingungen anregt. Die Frequenzkurve der Schwingungsamplitude liefert wichtige Aufschlüsse über die musikalischen Eigenschaften des Instruments. Bei der Auswertung beschränkt man sich auf eine objektive Beurteilung, die im wesentlichen die elastischen Schwingungen des Resonanzbodens betrifft. Weiter werden Ergebnisse einer großen Zahl von Versuchen an Geigen moderner oder älterer Konstruktion angegeben.

Summary. Electro-acoustical methods have been often used to study bowed-string instruments and, in particular at the O. M. Corbino Institute of Ultrasonics at Rome, the amplitude of vibration of the belly of such instruments, set in forced vibration by an S.H.M. of constant amplitude and varied frequency has been measured. An examination of the response characteristics leads to important conclusions about the musical quality of the instruments derived as purely objective judgements. Results are given from a large number of tests on modern and antique (restored) violins.

1. Plusieurs chercheurs ont déjà employé des méthodes électroacoustiques pour évaluer objectivement les instruments à archet et leur résultats principaux sont généralement bien connus [1] ··· [11].

À l'Institut d'Ultra-acoustique "Corbino", du Conseil National des Recherches (Rome), l'auteur a effectué des recherches systématiques sur les caisses harmoniques des violons, afin de remplacer le jugement artistique subjectif concernant la "bonté" de ces instruments par un jugement objectif, basé sur l'examen des courbes caractéristiques obtenues par des méthodes électroacoustiques.

La différence essentielle entre notre méthode de mesure et celle adoptée par d'autres chercheurs réside dans le type d'excitation des vibrations. En effet, on avait toujours fait vibrer le violon en agissant sur les cordes (à l'aide d'un "archet automatique"), et on avait mesuré l'intensité des sons émis aux différentes fréquences (analyse harmonique), ou bien on avait déterminé les différents modes des vibrations, du corps résonnant (relief des lignes nodales). Avec notre méthode, par contre, on excite directement la caisse harmonique, *sans cordes*, sur une petite aire centrale de la table d'harmonie (correspondant à la position habituelle du "chevalet"); la force vibromotrice, *sinusoidale*, est obtenue par excitation électrostatique, à l'aide d'une électrode à proximité de la surface à exciter qui a été métallisée.

On a fait varier progressivement la fréquence d'excitation de la caisse harmonique, et on a enregistré automatiquement l'amplitude de vibration en fonction de la fréquence (courbe de réponse).

Dans les courbes de réponse, les fréquences de résonance, auxquelles correspondent les amplitudes maxima de vibration, ont un intérêt évident, de même que la valeur moyenne de l'amplitude dans toute la gamme acoustique, ou dans une gamme de fréquences donnée.

L'auteur a déjà exposé le résultat de ces expériences [12], [13] qui ont prouvé la corrélation très étroite qui existe entre le jugement artistique des violons et l'allure des courbes de réponse. Nous avons montré, par exemple, que l'examen des courbes de réponse permet de suivre, et par conséquent de prédire, la dépendance des caractéristiques acoustiques d'un violon de la présence et de la position de *l'âme*. Par le simple examen des courbes de réponse on a aussi pu établir, entre certaines limites, si un violon possède les qualités fondamentales des instruments parfaits au point de vue acoustique (*puissance acoustique, équilibre, sensibilité, timbre*). Pour le timbre il faut tenir compte de la position des fréquences de résonance dans les gammes de fréquences choisies. Pour ces termes des définitions exactes de langage technique ont été données permettant une interprétation aisée du point de vue artistique.

Une série d'essais a établi l'importance fondamentale de la "table d'harmonie" a l'égard du fond et des éclisses; ceci se déduit en comparant les courbes de réponse des différentes parties du violon: table-fond-éclisses-touche, ou les courbes de réponse d'un même violon soit avec les éclisses et le fond d'érable, soit avec des éclisses en aluminium et un fond en plomb.

On a ainsi démontré que les épaisseurs des planches, et en particulier celles de la table d'harmonie, ont une grande importance vis à vis de la puissance acoustique émise. Contrairement à ce qui a été souvent affirmé, le vernis n'a pas du tout une influence considérable sur la puissance acoustique, mais il agit sur le timbre, c'est-à-dire sur la qualité du son.

2. D'après ces conclusions il est apparu évident que pour étudier le problème de la bonne construction des violons et de la restauration des instruments anciens, ce qui est d'un grand intérêt historique et artistique, on pouvait se borner essentiellement à analyser les caractéristiques physiques de la table d'harmonie. Dans ce but, on a construit un violon, qui a été jugé excellent, de la façon suivante: pour le fond et les éclisses on a employé du bois, de forme et d'épaisseurs traditionnelles, puisqu'il a déjà été montré que ces éléments ont une importance très petite. Par contre on a soigné la construction de la table d'harmonie, en suivant toutes ses modifications, à partir de la planche brute jusqu'à la table finie, à l'aide des courbes de réponse, enregistrées par la méthode susdite. Pour la table d'harmonie que par la suite nous appellerons I, les courbes de réponse ont été déterminées pour huit phases successives de la construction; quatre de ces courbes sont montrées dans la Fig. 1.

On est parti d'une petite table de bois de vieux sapin plus épaisse dans la zone centrale (18 mm), dont la forme était déjà celle d'un violon, mais encore à l'état brut. On a atteint progressivement la phase finale, et par conséquent des épaisseurs

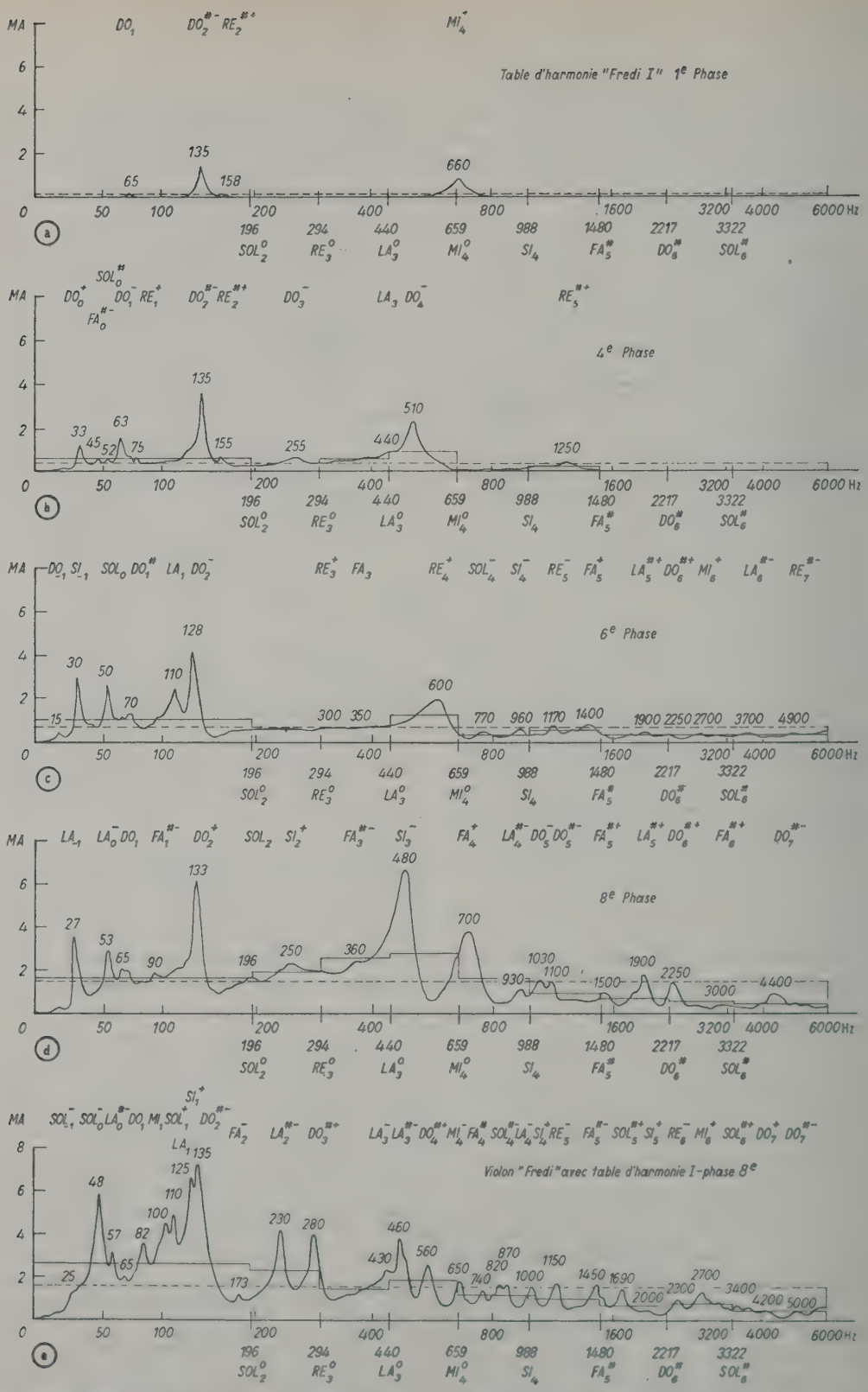


Fig. 1. Aspects différents de la courbe de réponse d'une table d'harmonie dans les phases successives de la confection:
(a) phase initiale (table rude, avec grosses épaisseurs de bois);
(b) et (c) phases intermédiaires: épaisseurs plus petites (sans chaîne, sans ouïes "ff");
(d) phase finale épaisseurs normales (avec chaîne et ouïes);
(e) courbe de réponse d'un violon "FREDI", avec la table d'harmonie I précédente (phase d).

normales (3,5 mm dans la région du chevalet et 2,5 mm vers les bords). Dans la phase finale la table d'harmonie a été garnie d'une chaîne et gravée avec les ouïes (ff) usuelles.

Il est évident, d'après la Fig. 1 a, b, c et d, que les courbes de réponse se modifient graduellement d'une phase à l'autre de la construction; c'est-à-dire, les fréquences de résonance de la table sont toujours plus nombreuses au fur et à mesure que la construction avance, et les amplitudes maxima aux résonances deviennent toujours plus grandes. L'amplitude moyenne de vibration (indiquée sur les figures par une ligne droite hachée) augmente, elle aussi, remarquablement.

La courbe de la Fig. 1e se rapporte à un violon complet, avec le fond et les éclisses ordinaires et la table d'harmonie I (caracterisée par la courbe de réponse 1d).

3. On a ensuite comparé plusieurs tables d'harmonie dans la phase finale de la construction, en relevant les courbes de réponse avant et après les avoir montées toujours sur le même fond et les mêmes éclisses.

Dans ce but on a choisi deux tables d'harmonie, indiquées par II et III, ayant à peu près les mêmes épaisseurs, mais étant différentes pour la qualité du bois, la chaîne et le mode de fixation de celle-ci.

Dans la Fig. 2a et 2c se trouvent les courbes de réponse de ces tables d'harmonie II et III; on voit qu'elles sont bien différentes entre elles et surtout qu'elles diffèrent beaucoup de la courbe de la table d'harmonie I (Fig. 1d).

Les courbes b et d de la même Fig. 2 se rapportent au violon complet obtenu en appliquant l'une ou l'autre des tables d'harmonie II et III sur le même fond et les mêmes éclisses (déjà employées pour la table d'harmonie I).

4. On a enfin enregistré les courbes de réponse de trois violons, ayant tous le même fond et les mêmes éclisses, auxquels on avait associé l'une des trois tables d'harmonie I, II ou III (Fig. 1e, 2b, 2d). (Le fond, les éclisses et les trois tables d'harmonie différentes ont été construits par M. le luthier FREDI de Rome.)

Le jugement artistique relatif aux trois violons a été obtenu par l'audition des mêmes morceaux musicaux, joués par le même artiste en présence d'un groupe de musiciens. La correspondance est parfaite entre l'évaluation artistique et celle à laquelle on est conduit si l'on compare les courbes de réponse. En d'autres termes le violon jugé le meilleur pour la puissance acoustique émise, l'équilibre, la sensibilité et le timbre est celui qui a la courbe de réponse la plus riche en fréquences de résonance, avec l'amplitude moyenne de vibration la plus élevée (Fig. 2d).

Il est encore important de remarquer que la courbe de réponse de la table d'harmonie seule appartenant au violon qui a été jugé le meilleur est celle qui est la plus riche en fréquences de résonance et dont l'amplitude moyenne est la plus élevée (comparer la Fig. 2c avec la Fig. 1d et la Fig. 2a).

Le bon vernissage des violons a coopéré favorablement à la qualité des sons.

5. Une étude tout à fait analogue a été faite également pour quelques cas de restauration de violons anciens et modernes. Dans ce cas aussi le fond et les éclisses ont été laissés inaltérés, tandis qu'on a substitué la chaîne, et parfois on a modifié légèrement l'épaisseur du bois de la table d'harmonie.

Pendant la restauration les courbes de réponse ont été enregistrées successivement dans les phases suivantes:

a) pour le violon original au complet (avant la restauration de la table d'harmonie);
b) pour la seule table d'harmonie (avec la chaîne originale);
c) pour la même table d'harmonie (avec une nouvelle chaîne plus convenable);
d) pour le violon constitué par le fond, les éclisses et le manche originels, complétés par la table d'harmonie restaurée.

Avant et après chaque restauration un jugement artistique a été émis, et la correspondance entre l'évaluation du caractère musical et celle objective déduite de l'examen des courbes de réponse, a été encore confirmée.

Parmi les nombreuses observations faites à l'occasion de la restauration d'une dizaine de violons, on peut mentionner les suivantes:

a) On constate que dans un violon bien restauré la courbe de réponse de la seule table d'harmonie, et celle du violon complet présentent des fréquences de résonance plus nombreuses que la courbe de la même table d'harmonie ou du même violon avant la restauration. Cela se verifie surtout dans la gamme comprise entre 400 Hz et 1500 Hz; ce qui confirme, encore une fois, l'importance tout a fait particulière de la table d'harmonie;

b) Pour tous les violons examinés on trouve des fréquences de résonance communes entre 50 Hz et 65 Hz, et 125 Hz ··· 135 Hz. Par contre sur les tables correspondantes étudiées isolément, qu'elles soient neuves ou restaurées, le maximum de réponse entre 50 Hz ··· 65 Hz est absent ou très faible, tandis que celui autour des 125 Hz ··· 135 Hz subsiste; cela montre que ce dernier est dû à la conformation particulière de la table tandis que le premier est dû,

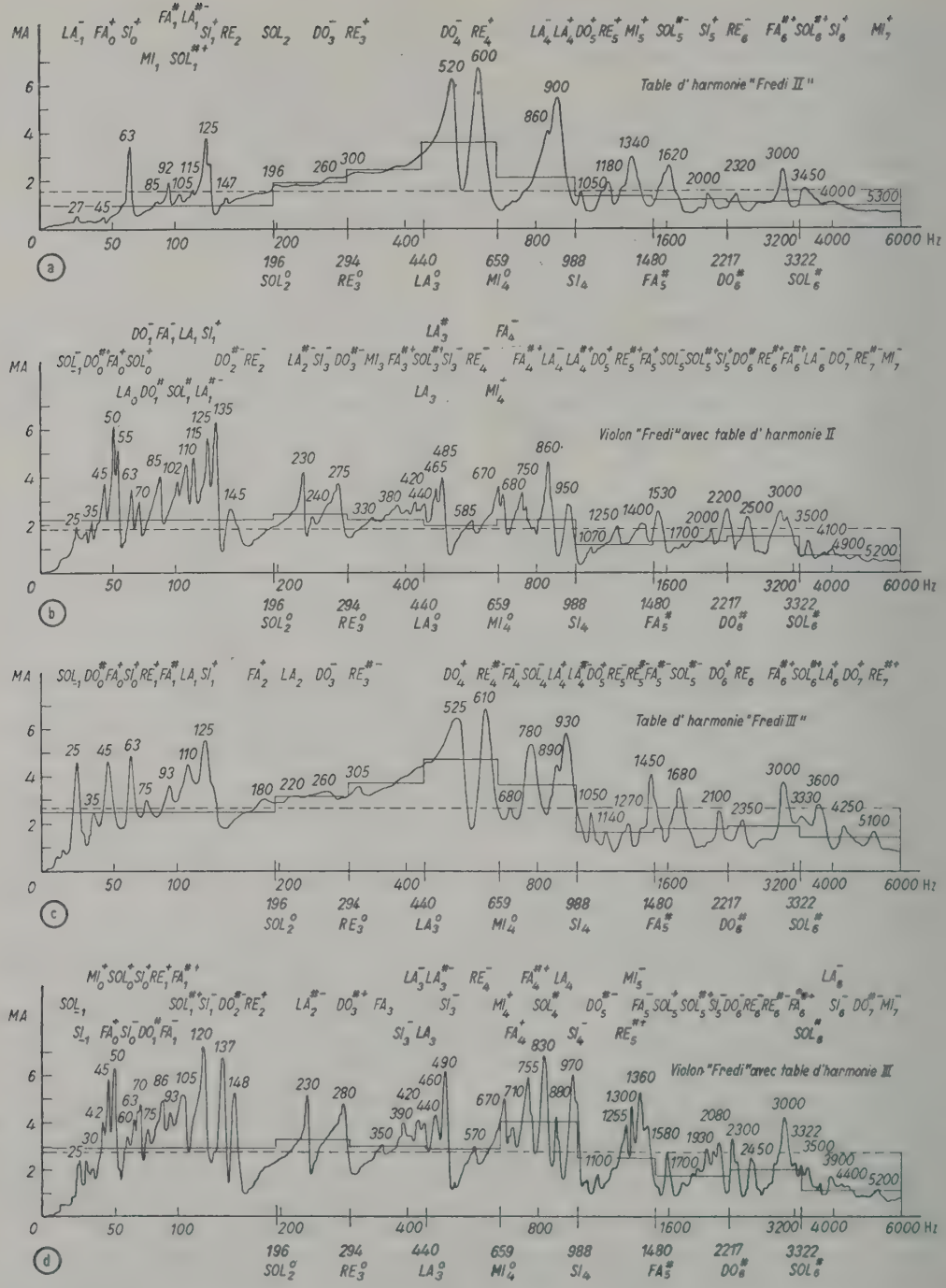


Fig. 2. Comparaison entre les courbes de réponse des tables d'harmonie et violons complets.
(a) table d'harmonie II;
(b) violon "FREDI" avec la table d'harmonie II (voir a);
(c) table d'harmonie III;
(d) violon "FREDI" avec la table d'harmonie III (voir c).

probablement, à l'air intérieur renfermé dans la caisse harmonique de l'instrument;

c) Lorsque les maximums de la courbe de réponse sont "aigus" ("coefficient de résonance" élevé pour le "mode de vibration" considéré), on constate que la durée de l'émission acoustique, dans la période transitoire, est la plus longue.

6. On s'est servi de la méthode précédente pour vérifier les propriétés élastiques du bois de sapin (vitesse du son, module d'YOUNG, frottement intérieur) et leur relation avec l'orientation des fibres, le vieillissement naturel, etc.

On a enfin effectué avec la collaboration de M. I. BARDUCCI [14], une recherche systématique sur des barrettes rectangulaires très minces (2 mm) excitées par flexion; les mesures ont été faites sur de nombreux types de bois, en particulier du bois de sapin utilisé pour la construction des tables d'harmonie.

Ces expériences semblent éclairer quelques aspects fondamentaux des phénomènes qui règlent les propriétés acoustiques des instruments à archet. Elles nous paraissent fournir une contribution utile à l'art de la lutherie.

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Discussion

RICHARDSON, E. G.: La présence de l'âme, et la forme et la position des ouïes, ont-elles une influence sur la courbe de réponse et sur la puissance acoustique du violon?

CONFÉRENCIER: L'allure de la courbe de réponse et la puissance acoustique du violon sont modifieés notablement si on enlève l'âme au violon ou encore si on agrandit ou si on ferme les ouïes.

MOLES, A.: Dans la méthode employée on opérait sur un violon sans cordes; ne croyez-vous pas que la tension des cordes modifie notablement les courbes de résonance en faisant participer le fond du violon par l'intermédiaire de l'âme à la résonance globale?

CONFÉRENCIER: En réalité la tension des cordes peut modifier les courbes de résonance globale du violon. J'ai l'intention d'étudier par la suite cette question, mais présentement pour des essais comparatifs il semble sage de ne pas introduire trop de variables, et de ne pas modifier la caisse harmonique du violon, l'âme et les conditions d'excitation.

MOLES, A.: Combien de sujets ont-ils contrôlé subjectivement les impressions de qualité et comment étaient faits les essais?

CONFÉRENCIER: La commission de contrôle subjectif était composée de 10 auditeurs (musiciens, techniciens, ouvriers). Le joueur, qui exécutait sur chaque violon le même morceau, était caché des auditeurs et il ne connaissait pas le violon joué ayant les jeux bandés.

BETRACHTUNGEN ZUM MUSIKALISCHEN ZUSAMMENKLANG

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Zusammenfassung. Wegen der beschränkten Zeitkonstanten und der nichtlinearen Verzerrungen hängt der Gehöreindruck auch von den Phasen der Teiltöne, d.h. vom Verlauf der Enveloppe der Schallvorgänge ab. Dieses Enveloppenhören ist nicht nur für Entfernungs- und Richtungseindruck wichtig, sondern spielt auch in der musikalischen Akustik eine entscheidende Rolle. Es gibt Anlaß zu zahlreichen akustischen Täuschungen. Durch das Enveloppenhören läßt sich auf Grund des Frequenzganges der inneren Reibung der Klanghölzer die Qualität der alten Geigen zwanglos deuten.

Summary. Owing to the limited time constant and the non-linear distortion of the human ear the sound impression does not only depend upon the amplitudes, but also upon the phases of the harmonics, i.e. on the envelope of the sound phenomena. This hearing of the envelope is not only important for the sensation of distance and direction, but also plays a fundamental role in musical acoustics. It is the cause of numerous acoustic illusions. The hearing of the envelope and the frequency dependence of the internal friction of timber used for musical instruments offer a natural explanation for the quality of old violins.

Sommaire. Du fait de la valeur finie de la constante de temps et du fait des distorsions nonlinéaires, la sensation sonore dépend non seulement des amplitudes, mais aussi des phases des har-

moniques, c'est à dire de l'enveloppe du phénomène sonore. Cette perception de l'enveloppe est non seulement importante pour les sensations de distance et de direction, mais joue aussi un rôle déterminant en acoustique musicale. Elle donne lieu à de nombreux effets d'illusion acoustique. On peut expliquer sans difficulté la qualité des anciens violons par cet effet d'enveloppe, en se basant sur la variation avec la fréquence du frottement interne des bois de lutherie.

Grundlagen der neuzeitlichen Akustik bilden die Fourierschen Theoreme und das Ohmsche Gesetz. Die Fourierschen Theoreme ermöglichen es, jeden Schallvorgang durch seine spektralen Komponenten darzustellen; das Ohmsche Gesetz, demzufolge das Ohr nur auf die Amplituden der Teiltöne, nicht aber auf ihre Phasen anspricht, erhebt diese spektralen Komponenten zur akustischen Realität und bildet so die Grundlage der Klanganalyse und der musikalisch-physiologischen Akustik. Dieses Ohmsche Gesetz, das einerseits einen großen Aufschwung der Akustik herbeiführte, blockierte andererseits lange Zeit jeden weiteren Fortschritt. Es stellt nämlich nur eine Näherung dar, die zwar bei hohen Frequenzen recht gut, bei tiefen Tönen aber überhaupt nicht mehr erfüllt ist. Sollte das Ohr eine ideale Fourieranalyse betreiben, so müßte es in der Lage sein, über die Grundperiode bei einmaligen Schallvorgängen, also über ein unendlich großes Zeitintervall zu integrieren, was offensichtlich unmöglich ist. Der Gehöreindruck wird daher im wesentlichen durch den momentanen Schwingungszustand bestimmt, wie er durch Fourierintegration über die Erinnerungszeit des Ohres, d.i. ungefähr eine zwanzigstel Sekunde, bewertet mit seiner Erinnerungsfunktion, gegeben ist. Diese beschränkte Integrationszeit des Ohres ermöglicht es, dem Verlauf der Enveloppe zu folgen und in Sprache und Musik kurzzeitige Klänge von Dauertönen zu trennen. Das Ohr hört also sehr viel mehr, als man in der Vergangenheit oft geneigt war anzunehmen; es hört auch die Phasen der Teiltöne, wie sie sich im Verlauf der Enveloppe der Schallvorgänge äußern.

Enveloppenmodulationen können musikalisch angenehme Wirkungen zur Folge haben, wenn sie, wie z.B. beim Vibrato, genügend langsam erfolgen, oder unangenehme, wie das Flatterecho, die Echovorgänge in schwach gedämpften Fernsprechleitungen oder das Schallfeld in akustisch ungünstigen Räumen, wenn ihre Periodizität etwa 16/s bis 60/s beträgt. Im Konzerthaussaal in Wien beispielsweise läßt sich durch bloße Unterteilung der bestehenden Reflexintervalle mittels eines oder zweier zusätzlicher Lautsprecher der außerordentlich harte Klangeindruck auf den seitlichen Galerieplätzen völlig beseitigen.

Als Folge des Enveloppenhörens empfindet das Ohr den besonderen Charakter des Toneinsatzes und Tonausklanges und auch die übrigen Ausgleichsvorgänge, die die stationären Dauertöne unserer Musikinstrumente ständig durchsetzen. Würde ein Klangbild nur streng stationäre Töne enthalten, so müßten sich die einzelnen Teiltöne zu einem Mischton überlagern und es wäre unmöglich, die verschiedenen Instrumente herauszuhören. Die Ausgleichsvorgänge dagegen sind für jedes Instrument charakteristisch und natürliche Unterscheidungsmerkmale; sie sind auch für Entfernungs- und Richtungseindruck maßgebend. Wie G. von Békésy [1] nachgewiesen hat, hängt der Entfernungseindruck wegen des Druck-Schnelleempfanges des menschlichen Ohres vom Krümmungsradius der Wellenfronten und damit von den tieffrequenten Ausgleichsvorgängen ab. Daß auch der Richtungseindruck durch die Ausgleichsvorgänge besonders prägnant übermittelt wird, bestätigen einfache Versuche. Der musikalisch-physiologische Wert dieses sogenannten räumlichen Hörens liegt aber nicht einmal so sehr in der räumlichen Breiten- und Tiefenwirkung und der durch sie bedingten Lebendigkeit des Klangbildes, man muß hören, wie der Schall im Orchester von einem Instrument zum anderen überspringt und muß hören, daß der Sänger im Vordergrund oder hinten steht. Ebenso empfindet man z.B. beim Orgelspiel das lebendige Hin- und Herschlüpfen des Klanges zwischen den einzelnen Orgelpfeifen als reizvoll und für eine gute Klangwirkung unerläßlich. Die Ausgleichsvorgänge erst entscheiden die musikalisch plastische Wirkung eines Klangbildes, lassen die Gattung des Einzelinstrumentes erkennen und es als solches hervortreten (Backhaus [2]). Beobachtet man z.B. den Verlauf des Schalldruckes beim Anstreichen einer Geige, so sieht man am Schirmbild der Braunschen Röhre die Maxima und die relativen Phasen zwischen den Teiltönen ständigen kleinen Schwankungen unterworfen. Das Oszillographenbild atmet und lebt und dieses „Leben" ist es offenbar, das den musikalischen Klang vor dem eines gewöhnlichen Tongenerators auszeichnet. Im übrigen ist bemerkenswert, daß die tieffrequenten Ausgleichsvorgänge ebenso beim Spielen hoher wie tiefer Töne angeregt werden, so daß z.B. eine Geige, deren G-Saite schlecht klingt, auch beim Spiel hoher Töne minderwertig wirkt, was den Musikern wohl bekannt ist.

Werden die tieffrequenten Ausgleichsvorgänge unterdrückt, so verliert das Klangbild seine räumliche Tiefe und wirkt fad und dünn, wie z.B. die Wiedergabe eines zu leise eingestellten Radioapparates oder eine Musikdarbietung im Freien oder in einer überdämpften Halle (wo die tieffrequenten Vorgänge durch die stärkere Abnahme der Ohremp-

findlichkeit bei tiefen Frequenzen für schwache Amplituden früher unter die Hörschwelle fallen). Nur wenn das Orchester durch Instrumente mit kräftigen Ausgleichsvorgängen, wie Trompeten und Trommeln und Pauken, ergänzt wird, können dann reizvolle Klangwirkungen erzielt werden.

Auch in großen Räumen kommt wegen des großen Abstandes der Schallquellen das eigentliche Entfernungs- und Richtungshören wenig zur Geltung und das Klangbild muß durch künstliche Maßnahmen, sozusagen durch Tricks, belebt werden. Da das Ohr relativ primitiv ist, lassen sich nämlich die tieffrequenten Ausgleichsvorgänge teilweise durch tieffrequente Vorgänge verstärken. So kommt man zu der Forderung, daß umso mehr tief gestimmte Instrumente im Orchester mitwirken müssen, je größer der Raum, und daß eine Kirchenorgel umso längere Pfeifen haben soll, je größer das Kirchenschiff ist. Da ferner die Ausgleichsvorgänge für das Heraushören der einzelnen Instrumente entscheidend sind, und der erste Reflex wegen der Kugelwellenausbreitung und der physiologischen Vorgänge immer stärker wirkt als die nachfolgenden [3], [4], werden die vorderen Instrumente immer besser gehört als die rückwärtigen, obwohl die objektiv gemessene Lautstärke im Saal praktisch nicht von der Gruppierung abhängt.

Der musikalisch physiologische Wert der Ausgleichsvorgänge, besonders der tieffrequenten, wird dadurch noch wesentlich gesteigert, daß sie lauter gehört werden, als sie ihren Amplituden nach bewertet werden dürften. Möglicherweise ist es das ständige Training des Ohres, das den Ausgleichsvorgängen als den charakteristischen Erkennungszeichen und als den Entfernungs- und Richtungsmarken der Schallquellen gesteigerte Aufmerksamkeit zollt, möglicherweise ist es aber die erhöhte Zahl der Nervenimpulse, die dem Gehirn unmittelbar nach Anschalten eines Tones zugeführt wird [5] oder andere physiologische Erscheinungen. Läßt man etwa zwei Stimmgabeln der Frequenzen 3000 und 3300 Hz gleichzeitig ertönen, so hört man einen Differenzton, der die Lautstärke der Primärtöne zu erreichen, ja sogar zu übertreffen scheint. Dieser Differenzton bestimmt nach kurzer Einwirkung auf das Ohr sogar die Tonhöhe des Zusammenklanges. Trotzdem beträgt seine Amplitude, wie u.a. W. Kuhl [6] nachgewiesen hat, nur etwa 1,5 % der Primärtöne. Beim Messen der Amplitude des im Ohr erzeugten Differenztones nach der Schwebungsmethode wirkt der Meßton unter gleichen Bedingungen wie der Differenzton, so daß wohl der wahre Wert der Amplitude, nicht aber der subjektive Lautstärkeeindruck bestimmt wird. Das Ohr verstärkt demnach die Amplitude des Differenztones um etwa 30 dB. Eine Reihe ähnlicher Wirkungen sind aus dem täglichen Leben wohl bekannt: Beim Postmikrophon beispielsweise hört man immer den Grundton der Sprache, obwohl er in der Regel gar nicht vorhanden ist; der Heulton einer Sirene wirkt immer eindringlicher als ihr stationärer Klang usw., das Vibrato hebt den Ton eines Instrumentes aus dem Orchester hervor. Offenbar handelt es sich hier um akustische Täuschungen, deren Analoga in der Optik wohl bekannt sind. Betrachten wir beispielsweise zwei genau gleich groß gezeichnete Würfel stereoskopisch, so werden sie sehr groß oder sehr klein gesehen, je nachdem, ob ihr Fluchtpunkt so liegt, daß sie unmittelbar vor dem Auge oder in weiter Ferne erscheinen. (Beim Stimmgabelversuch ist es der tieffrequente Differenzton, der in unmittelbarer Nähe des Ohres lokalisiert wird und entsprechend kräftiger wirkt, während die Primärtöne scheinbar aus weiter Ferne herangetragen werden.) Die Zeitschrift „Life" veröffentlichte vor einiger Zeit eine Reihe verschieden großer Köpfe, die einer Kugel gegenübergestellt werden. Bringt man die Kugel mit dem großen Kopf in Verbindung, so handelt es sich offenbar um einen kleinen Ping-Pong Ball, sucht man dagegen eine Beziehung zum kleinsten der Köpfe, so erscheint sie als mannsgroßer Ball. Ein besonders schönes Beispiel ist eine mit einem Butzenscheibenmuster beklebte Fensterscheibe. Akkomodiert man das Auge, 50 cm vor der Scheibe, durch ein Loch im Muster auf die Ferne, so erscheint das Muster plötzlich in einem Abstand von etwa 2 m riesengroß. Wir sehen also einen Gegenstand groß oder klein, je nachdem mit welchen Objekten wir ihn in Verbindung bringen und wie wir das Auge akkomodieren. Es darf nicht überraschen, wenn das Ohr, das primitiver ist als das Auge, ähnlichen Täuschungen unterliegt. Schließlich lassen die vorangegangenen Betrachtungen auch die Bedeutung des plastischen Hörens für die Qualität der Wiedergabe in anderem Licht erscheinen. Genau so, wie optisch die natürliche Farbe, das Funkeln und Glitzern der Gegenstände nur bei stereoskopischer Betrachtung zur Geltung kommen, bleibt der Reiz eines natürlichen Klanges nur bei räumlicher Wiedergabe erhalten. Während wir beim einohrigen Hören und einäugigen Sehen von Natur aus bestrebt und in der Lage sind, aus der Bewegung der beobachteten Gegenstände oder durch Bewegung des Auges oder des Kopfes die fehlende Plastik eines unmittelbaren Sinneseindruckes zu regenerieren, können wir bei der Wiedergabe toter Gebilde, sofern sie annähernd naturgetreu sein soll, auf die echte Plastik nicht verzichten.

Durch die vorangegangenen Betrachtungen zeichnet sich die Bedeutung des Klirrfaktors für Übertragungsanlagen ab: Während er bei der Wieder-

gabe von Soloinstrumenten offenbar relativ unwichtig ist, da er hier lediglich auf die Klangfarbe Einfluß nimmt, könnte sich ein Klirrfaktor in der Übertragungsanlage bei Orchesterdarbietungen äußerst störend auswirken, da die Differenztöne, wie aus dem Stimmgabelversuch hervorgeht, unter Umständen 30 dB lauter gehört werden, als sie wirklich sind. Im allgemeinen sind hier die Verhältnisse jedoch noch ungeklärt und weitere Versuche erforderlich.

Mit dem gesamten eben erörterten Fragenkomplex steht das umstrittene Problem der alten Geigen in enger Berührung. Abb. 1 zeigt den Frequenzgang der inneren Reibung verschieden alter Klangfichten, wie sie im Geigenbau Verwendung finden, und einer Baufichte. Bemerkenswert sind die geringe Dämpfung des gealterten Holzes bei tiefen Frequenzen und die durch die kräftigen tieffrequenten Ausgleichsvorgänge bedingte Tragfähigkeit der alten Instrumente, sowie die beträchtliche innere Reibung

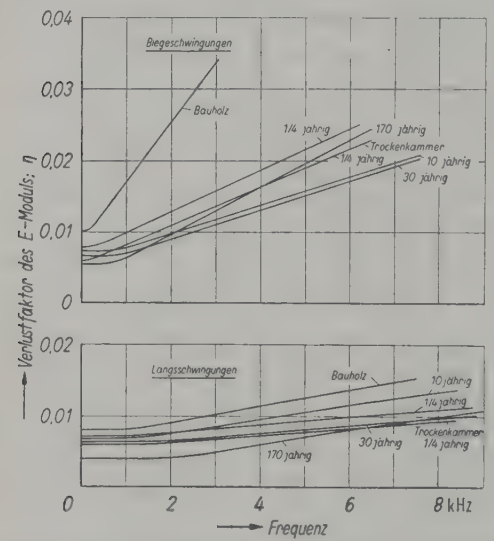


Abb. 1. Die innere Reibung von Klangholz für Transversal- und Biegeschwingungen (Diplomarbeit PTACNIK, T. H. Wien).

bei hohen Frequenzen, die einen weichen Klang zur Folge hat. Neue Geigen sind bei tiefen Frequenzen stärker gedämpft und daher in der Regel weniger tragfähig, klingen aber wegen der geringeren Dämpfung bei hohen Frequenzen etwas schärfer und oft frischer. Durch Lackieren wird die Dämpfung jungen Holzes erhöht; der Klang lackierter neuer Geigen nähert sich daher dem alter Instrumente. Die schlechtere Tragfähigkeit neuer Geigen dagegen wird durch den Lack nicht verbessert. Da die innere Reibung von Holz bei hohen Frequenzen mit zunehmendem Alter zunimmt, verliert der Lack mit dem Alter der Geige an Bedeutung. Auch eine Zunahme an Feuchtigkeit oder Temperatur hat bei hohen Frequenzen ein Weicherwerden des Klanges zur Folge, während die hysteresebedingte Reibung bei tiefen Frequenzen nicht beeinflußt wird. Das „Warmspielen" oder Erwärmen von Instrumenten vor dem Konzert ist also, besonders wenn sie neu sind, kein leerer Wahn. Es ist in diesem Zusammenhang nicht uninteressant, daß auch Glockenlegierungen eine starke Zunahme der inneren Reibung nach hohen Frequenzen zu aufweisen, wie am Beispiel der Pummerin in Wien nachgewiesen werden konnte und daß Kunststoffe, die einen konträren Verlauf der inneren Reibung erkennen lassen, für den Musikinstrumentenbau ungeeignet sind.

Einige Bemerkungen seien in Hinblick auf eine in der Acustica erschienene Arbeit [7] angeschlossen:

Bei Musikinstrumenten, wie z.B. Geigen, ist der Klirrfaktor bei geringen Lautstärken in der Regel klein, erreicht aber bei 100 Phon 20 % und mehr (Abb. 2, Abb. 3). Regt man die Geige in einem Minimum der Frequenzkurve sinusförmig mit einer Tauchspule an, so gelingt es oft mit 40 Watt Erregung, kaum größere Lautstärken als 60 bis 80 Phon zu erzielen. Man kann dann sogar erreichen, daß der Grundton überhaupt fehlt und nur die doppelte Frequenz abgestrahlt wird. Aber nicht nur der Klirrfaktor, sondern auch schlechte Stellen der Frequenzkurve bewirken ein Überhandnehmen der Obertöne und damit einen rauhen Klangeindruck. Die Ausgeglichenheit der Frequenzkurve und die technisch einwandfreie verzerrungsarme Konstruktion sind nach Auffassung des Verfassers wichtige

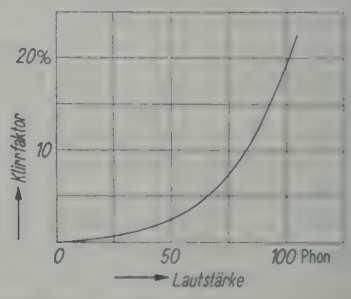


Abb. 2. Die Abhängigkeit des Klirrfaktors von der Lautstärke in der Nähe einer Einbuchtung der Frequenzkurve.

Gründe für die Qualität einer Geige. Diskrepanzen, wie sie in der genannten Arbeit und den Auffassungen des Verfassers zum Ausdruck kommen, gehen wohl im wesentlichen auf das Fehlen einer einwandfreien Definition des Klirrfaktors zurück. Man muß jedenfalls festsetzen, ob man bei der Messung des Klirrfaktors die Lautstärke konstant hält, oder den Er-

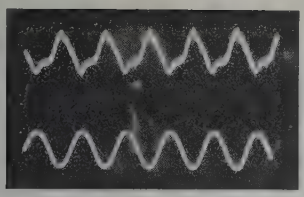


Abb. 3. Der Schalldruck in 30 cm Abstand von einer Geige bei elektrodynamischer Erregung des Steges in der Nähe eines Minimums der Frequenzkurve.

regerstrom und darf dabei nicht übersehen, daß man bei schlechteren Geigen in den Minima der Frequenzkurve selbst beim Durchbrennen der Erregerspulen kaum 80 Phon erreicht. Auf einen häufigen Trugschluß sei besonders hingewiesen: Kleine Amplituden bedeuten nicht notwendigerweise einen kleinen Klirrfaktor, was am Beispiel des schallharten Kohlemikrophons besonders deutlich wird.

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STEREOPHONISCHE ÜBERTRAGUNG KLANGSCHWACHER INSTRUMENTE IM KONZERTSAAL

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Zusammenfassung. Der Klang bestimmter Instrumente kann unter Umständen im Konzertsaal schwächer hörbar sein als erwünscht. Gründe: 1. der schwache Klang ist eine Eigenart des Instrumentes, 2. das Instrument steht an einem ungünstigen Platz. In beiden Fällen kann der Klang einer stereophonischen Wiedergabe dem Originalschall beigemischt werden. An drei Beispielen werden die elektroakustischen Hilfsmittel erläutert, die so anzuwenden sind, daß die natürliche Klangwirkung erhalten bleibt und der Hörer nichts von der Mitwirkung der Übertragungsanlage bemerkt.

Summary. The sound of certain instruments can sound weaker in the concert hall than desired. This may be due to the weak sound being a property of the instrument or the instrument being in an unfavourable position. In both cases the sound can be mixed with a stereophonic reproduction of the original. Three examples are given of the use of an electro-acoustic device that does not change the natural timbre, while the listener is unaware of the use of the reproducing apparatus.

Sommaire. Dans certaines circonstances, la sonorité d'instruments de musique peut sembler plus faible dans une salle de concert qu'on ne le désirerait. Cela peut être dû à la nature propre de l'instrument ou à une position défavorable de cet instrument. Dans ces deux cas, on peut ajouter à la sonorité propre celle d'une reproduction stéréophonique. On montre sur 3 exemples comment employer les dispositifs électro-acoustiques auxiliaires pour que soit conservé le caractère propre du son et que l'auditeur ne se doute pas de l'adjonction du dispositif de transmission.

Das Prinzip der Stereophonie ist — als Kopfhörerübertragung — vor über 70 Jahren durch C. Ader in Paris bekannt geworden [1]. Auf theoretischen Arbeiten von Helmholtz [2] aufbauend, haben Fletcher und seine Mitarbeiter später in den Jahren 1930 ··· 1934 das Verfahren der stereophonischen Lautsprecherübertragung entwickelt [3]. Obwohl der bedeutende klangliche Gewinn, den die Wiedergabe einer natürlich wirkenden Hörperspektive mit sich bringt, seither in zahlreichen überzeugenden Versuchen gezeigt werden konnte [4], hat die Stereophonie bis heute noch keine allgemeine Verbreitung gefunden. Das liegt daran, daß die wirtschaftlichen Erwägungen, die in der Praxis der Schallplatten-, Band- oder Tonfilmwiedergabe stets im Vordergrund stehen, den vermehrten Aufwand von zwei oder drei voneinander unabhängigen Übertragungskanälen kaum tragbar erscheinen lassen. Besonders aber der Rundfunk, der als Mäzen für die Elektroakustik auch die Stereophonie entscheidend fördern könnte, verhält sich — zumindest in Deutschland — betont zurückhaltend, weil eine einwandfreie Rundfunk-Fernübertragung von zwei Kanälen erhebliche hochfrequenztechnische Schwierigkeiten verursacht *).

Eine vorteilhafte und leicht realisierbare Anwendung der Stereophonie ergibt sich jedoch bei be-

*) Wie neuerdings bekannt wird, ist der amerikanische Rundfunk demgegenüber sehr interessiert an der Stereophonie.

stimmten akustischen Problemen, die bei musikalischen Veranstaltungen in großen Konzertsälen und Kirchen oder auch im Opernhaus auftreten können:

1. Die Verstärkung eines klangschwachen Instrumentes, um es in einem größeren Raum, insbesondere im Vergleich zu anderen Instrumenten, in angemessener Stärke zum Klingen zu bringen (Cembalo).

2. Die Überwindung der klanglichen Nachteile, die sich durch die ungünstige Position eines Klangkörpers im Raum ergeben können (Orgel, Chor, verdeckte Teile des Opernorchesters).

Wesentliche Voraussetzung für die Lösung dieser Aufgaben ist eine vom künstlerischen Standpunkt aus vollkommen einwandfreie und natürlich klingende Wiedergabe, derart, daß auch der anspruchsvolle, kritische Zuhörer die Mitwirkung elektroakustischer Übertragungsmittel nicht bemerkt. Das ist durch die folgenden Maßnahmen erreichbar:

a) Die verwendeten Mikrophone, Verstärker und Lautsprecher müssen höchste Qualität besitzen.

b) Alle Veränderungen in der Klangfarbe, die durch die im Nahfeld angeordneten Mikrophone und durch eine vergrößerte Schallstärke (unter Berücksichtigung der Ohreigenschaften) hervorgerufen werden, müssen mit Hilfe geeigneter Entzerrungsmaßnahmen wieder rückgängig gemacht werden.

c) Der von den Lautsprechern abgestrahlte Schall muß dem Originalschall beigemischt werden. Die Verstärkung soll gering sein und jedenfalls nicht größer, als es zur Erzielung eines ausgewogenen Klangeindruckes im Raum unbedingt nötig ist.

d) Die Lautsprecher müssen so angeordnet werden, daß der erste Schalleindruck den Hörer jeweils von *dem* Ort her erreicht, an dem er den verstärkten Klang wahrnehmen soll [5]. Die Schallwegdifferenz zwischen Original und Lautsprecher soll nicht mehr als etwa 10 m betragen.

e) Durch Verwendung von gerichteten Mikrophonen und Lautsprechern müssen akustische Rückkopplungen verhindert werden.

f) Zur Erzielung einer natürlichen Raumwirkung muß eine Mehrkanalübertragung durchgeführt werden.

In Abb. 1 ist eine Apparatur zur Schallverstärkung des Cembalos mit dem zugehörigen Frequenzgang dargestellt. Die Mikrophone befinden sich unmittelbar über dem Resonanzboden an zwei klanglich möglichst unterschiedlichen Plätzen, M_I in der Diskantzone, M_{II} in der Baßzone. Der starke Abfall der Frequenzkurve im tiefen Frequenzgebiet ist notwendig, um a) die verstärkte Wirkung der tiefen Frequenzen im Nahfeld der Schallabstrahlung und b) die größere Empfindlichkeit des Ohres für tiefe Frequenzen bei steigender Lautstärke auszugleichen. Wie die Versuche gezeigt haben, ist jedoch auch noch eine Reduzierung des Übertragungsmaßes im Gebiet zwischen 1000 und 2000 Hz erforderlich. Ohne diese Entzerrung wirkt der Klang des Cembalos derb und unedel, sobald man die Lautstärke über diejenige des Originals hinaus anwachsen läßt. Offensichtlich handelt es sich hier um eine noch nicht recht geklärte psychologische Erscheinung. Die Tatsache, daß die sogenannten Näselformanten im Bereich von 1000 ··· 2000 Hz liegen, und daß andererseits wertvolle Geigen, wie BACKHAUS [6] und MEINEL [7] gefunden haben, zwischen 1000 und 2000 Hz verhältnismäßig wenig Teiltöne aufweisen, läßt vermuten, daß das Ohr — ästhetisch gesehen — eine gewisse Abneigung gegen dieses Frequenzgebiet hat. Möglicherweise kommt dieser Beobachtung eine grundsätzliche Bedeutung zu.

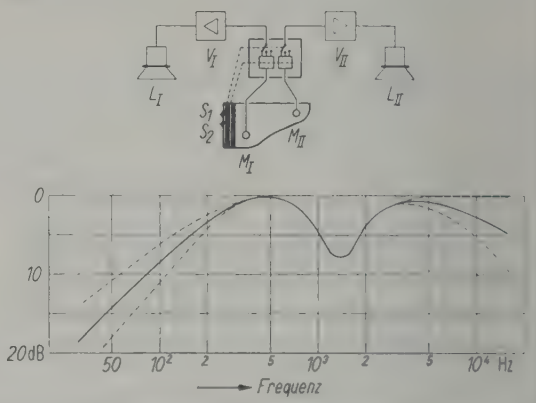


Abb. 1. Prinzipschaltbild und Frequenzgang der Apparatur zur Schallverstärkung des Cembalos.

Der Schalter S_1 dient zur Veränderung der Gesamtverstärkung um 2 Stufen von je 3 ··· 4 dB. Mit dem Schalter S_2 können die hohen und tiefen Frequenzen in geringen Grenzen variiert und damit der Klang der Lautsprecherwiedergabe den besonderen Verhältnissen des Raumes noch besser angepaßt werden (vgl. die gestrichelten Kurvenzüge). Die Lautsprecher müssen etwas weiter von den Zuhörern entfernt stehen als das Cembalo, damit der erste Schalleindruck durch den Originalschall entsteht und die Hörer ihn auf das Instrument selber und nicht auf die Lautsprecher lokalisieren.

Mit Hilfe dieser Apparatur läßt sich der Cembaloklang auch in einem größeren Saal einem Kammerorchester äquivalent gegenüber stellen. Ein solches Konzert ist zum ersten Mal im Jahre 1941 in Berlin veranstaltet worden [8]. Da wegen des Krieges außerhalb Deutschlands über diese Versuche wenig bekannt geworden sein dürfte, schien es angebracht, hier noch einmal darüber zu berichten.

Auch der Flügel oder die Orgel beispielsweise lassen sich in extrem großen Sälen durch Verwen-

dung einer derartigen Übertragungsapparatur viel klangschöner zur Wirkung bringen, als wenn man durch konstruktive Maßnahmen an den Instrumenten deren Schallintensität über das übliche Maß hinaus steigern würde, beim Flügel etwa durch noch dickere Saiten und schwerere Hämmer, bei der Orgel durch höheren Winddruck. Denn der Schallintensität der Musikinstrumente sind natürliche Grenzen gesetzt, die nur unter Verlust ihrer ästhetisch befriedigenden Klangwirkung überschritten werden können. Die Übertragungsanlage hat also in allen diesen Fällen die Funktion einer *Leistungsanpassung* des Klangkörpers an den Raum.

Das zweite Problem ist die akustische Verbesserung der klanglichen Nachteile, die durch eine ungünstige Aufstellung einer Orgel oder von Teilen eines Orchesters hervorgerufen werden. Aus Abb. 2 ist ersichtlich, wie der Klang einer Orgel, die hinter dem Orchesterpodium in eine Nische in der Rückwand eines Konzertsaales eingebaut ist, mit Hilfe einer stereophonischen Anlage gleichsam nach vorne geholt werden kann, so als ob die Orgel auf gleicher Höhe mit den vorderen Teilen des Orchesters stände. Im Hinblick auf die Ausdehnung der Orgel empfiehlt sich eine Übertragung mit drei Kanälen. Dabei genügt es, wenn vorwiegend die für die Lokalisierung wichtigen hohen und mittleren Frequenzen übertragen werden. Die Abschwächung der Wiedergabe im tiefen Frequenzgebiet bedeutet zugleich eine praktische Erleichterung sowohl in bezug auf die Lautsprechergröße als auch für die Entkopplung zwischen den Lautsprechern und den Mikrophonen.

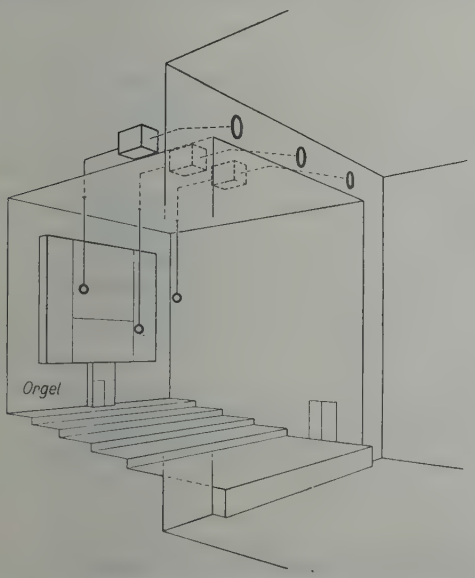


Abb. 2. Stereophonische Übertragung einer ungünstig stehenden Orgel im Konzertsaal.

Durch den Laufzeitunterschied zwischen der verfrühten Wiedergabe und dem Originalschall entsteht eine gewisse Hallwirkung, die dem Orgelklang in jedem Falle zuträglich ist, ganz besonders in einem gefüllten Konzertsaal, dessen Nachhallzeit in der Regel sowieso zu klein ist, um Orgelmusik gut zur Geltung kommen zu lassen. Eine Anlage dieser Art ist unlängst in der Hamburger Musikhalle erprobt worden. Dabei konnte gezeigt werden, daß die klanglichen Nachteile, die die Orgel durch ihre rückwärtige Lage und durch den geringen Nachhall des Raumes erleidet, mit Hilfe einer stereophonischen Übertragung entsprechend der Abb. 2 weitgehend behoben werden können.

Abschließend sei noch auf die Möglichkeit hingewiesen, in ähnlicher Weise das Problem des verdeckten Opernorchesters zu lösen. Wie Abb. 3 zeigt, läßt sich der Klang der verdeckten Teile des Orchesters (im wesentlichen die Bläser und Schlaginstrumente) von den in die Rampe eingebauten Lautsprechern günstig abstrahlen, wodurch das klangliche Gleichgewicht zwischen Streichern und Bläsern im Raum leicht hergestellt werden kann. Durch Verwendung von Gradientmikrophonen in der dargestellten Anordnung sind Rückkopplungen trotz der geringen Mikrophon-Lautsprecher-Entfernung leicht vermeidbar. Auch hier kann man sich im wesentlichen auf die Übertragung der mittleren und hohen Frequenzen beschränken.

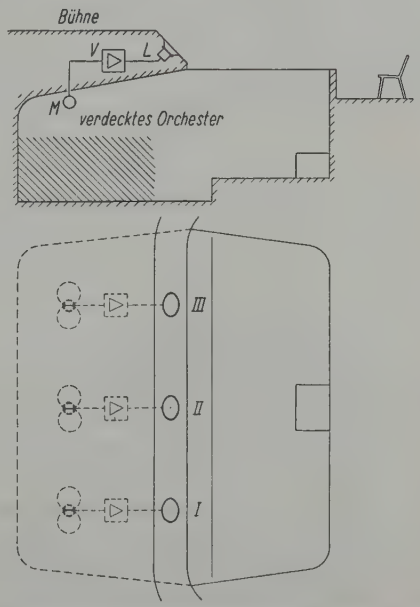


Abb. 3. Schallübertragung eines verdeckten Opernorchesters.

Durch die eingangs erwähnte Voraussetzung, daß die Mitwirkung der Lautsprecheranlage dem Hörer nicht bewußt werden soll, daß also die Übertra-

gungsapparaturen nicht klanglicher Selbstzweck sein dürfen, sondern Hilfsmittel bleiben und sich jederzeit den künstlerischen Forderungen unterordnen müssen, unterscheiden sich die beschriebenen Anwendungen der Stereophonie grundsätzlich von jenen Verfahren, die allgemein als „elektrische Musikinstrumente" oder — besser — als „elektronische Klangerzeugung" bezeichnet werden. In einer Zeit, in der die Frage, wie weit man der Technik beherrschenden Einfluß auf die künstlerische Gestaltung gewähren darf, in wachsendem Maße diskutiert und von den Künstlern und Philosophen oft genug mit großer Skepsis gegenüber den technischen Errungenschaften beantwortet wird [9], darf man die hier erörterten Beispiele wohl als positive Beiträge der Elektroakustik zum Musikleben werten. Sie wollen in einem analogen Sinne verstanden werden wie etwa die Rundfunkübertragung oder das Schallplattenverfahren, deren Hauptaufgabe die musikalische Denkmalspflege geworden ist. In der Tat ist es bei dem gegenwärtigen Stand der Elektroakustik weniger ein physikalisches oder technisches Problem, wie derartige Übertragungsverfahren angewendet werden, sondern vielmehr in erster Linie ein *künstlerisches.*

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Diskussion

Winckel, F.: Die Einsenkung in der Frequenzkurve zwischen 1000 und 2000 Hz (siehe Abb. 1) habe ich ebenso bei der Untersuchung des Timbre von Gesangstimmen festgestellt. Es ist ein Qualitätsmerkmal hervorragender Stimmen. Die Erklärung dafür liegt darin, daß der ä-Formant, der eng mit der Nasalität zusammenhängt, in dem Bereich von 1000 ··· 2000 Hz dem Klang einen etwas quäkenden Charakter gibt und damit musikalisch-ästhetisch zu beanstanden ist.

Vortragender: Ich möchte der Bemerkung von Herrn Winckel voll und ganz zustimmen. Diesen Frequenzgang habe ich seinerzeit bei der Erprobung der Cembalo-Apparatur rein gehörsmäßig gefunden. Daß ähnliche Beobachtungen nicht nur bei Geigen sondern auch beim Gesang gemacht werden, scheint die Vermutung zu bestätigen, daß wir es hier mit einer allgemeingültigen Feststellung über das musikalisch-akustische Qualitätsempfinden zu tun haben.

ELEKTROAKUSTISCHE MITTEL IN DER AKTIVEN TONKUNST

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Zusammenfassung. Auf Grund der Gehöreigenschaften werden Richtlinien für die Entwicklung neuer Musikinstrumente aufgestellt. Als hauptsächliche Schwäche der althergebrachten Instrumentalmusik wird das für Melodik und Harmonik gemeinsame Intervallsystem bezeichnet. Freiheit der Melodikbildung durch den Spieler und strenge Harmonik durch Frequenzdivision wird vorgeschlagen. Stärker differenzierte Klangfarben sollen durch Erzeugung beliebiger Schwingungen gebildet werden, welche durch eine Synchronisationsmethode phasenrichtig in jede Periode des Grundtones eingeblendet werden.

Summary. Indications are given, based on hearing characteristics, of the way new developments in musical instruments should go. The commonly employed scale is shown as a principal weakness for melody and harmony on old instrument types. More freedom for the player in melody and strong harmony by frequency division is required. More sharply differentiated timbres can be built up from certain vibrations which by a process of synchronisation can be blended in correct phase relationship with each period of the fundamental.

Sommaire. On établit, à partir des propriétés caractéristiques de l'ouïe, les règles à suivre dans la conception de nouveaux instruments de musique. La principale faiblesse de la musique instrumentale composée jusqu'ici est due à l'emploi d'un système d'intervalles commun pour la mélodie et l'harmonie. On préconise de rendre à l'éxécutant la liberté de conception de la mélodie et d'adopter une

harmonie rigoureuse par subdivision des intervalles de fréquence. On obtiendra des timbres fortement différenciés en produisant des oscillations au choix de l'exécutant, et en les introduisant avec la phase voulue à chaque période du son fondamental, grâce à une méthode de synchronisation.

Die gigantische Entwicklung der Technik in unserem Zeitalter hat vor dem Musikinstrumentenbau Halt gemacht, obschon dieser ohne Zweifel ein Gebiet der Technik ist. Wohl hat die Speicherung und Übertragung von Musik eine große technisch-wirtschaftliche Entwicklung genommen, aber die Technik ist damit nicht in die aktive Tonkunst eingedrungen. Das Mikrophon ist der eiserne Vorhang, der die Technik von der Musik trennt.

Elektroakustische Musikinstrumente konnten bis heute noch keine größere Bedeutung in der Tonkunst erlangen. Über diese Tatsache können auch gewisse wirtschaftliche Erfolge nicht hinwegtäuschen. Die elektroakustischen Musikinstrumente, welche eine Breitenwirkung erzielen konnten, haben sich entweder auf surrogative Aufgaben oder auf die Unterhaltungsmusik beschränkt. Das Instrument des Verfassers hat im Bereich der seriösen Musik große Erfolge errungen, die einem vorläufig noch kleinen Kreis von Tonkünstlern zu verdanken sind [1].

Der Konstrukteur von neuen Musikinstrumenten steht hinsichtlich der Einfühlung in den Gebrauchszweck und in die Wünsche der Konsumenten sowie der Voraussicht künftiger Entwicklung vor ungleich schwierigeren Aufgaben als auf anderen Gebieten der Technik. Große Tonkünstler können ihn wohl in der Richtigkeit seines Strebens bestärken, die Musikgeschichte kann ihn lehren, daß sich die Tonkunst in Richtung einer Intensivierung und stärkeren Differenzierung der musikalischen Ausdrucksformen fortentwickelt, aber konkrete Richtlinien wird er nicht erwarten dürfen.

Die Elektronik bietet für Musikinstrumente so universelle Konstruktionsgrundlagen, daß es richtiger ist, das Problem als ein völlig neues anzusehen, statt sich die althergebrachten Instrumente zum Vorbild zu nehmen oder sich einem speziellen Musikstil zu verpflichten.

Die naturwissenschaftliche Grundlage der Musik ist die Schallwahrnehmung durch das Gehör. Allenfalls könnte man das Gefühl für den Rhythmus als außerhalb der Hörempfindung liegend annehmen. Die Tonkunst scheidet aus den wahrnehmbaren Schallvorgängen die primitiven als für sie ungeeignet aus, nämlich die sinusförmigen Schwingungen und das weiße Rauschen. Was dazwischen liegt, soll von den Musikinstrumenten darstellbar sein.

Damit soll nicht gesagt sein, daß ein elektronisches Musikinstrument ein akustisches Laboratorium sein müsse. Wir müssen vielmehr die für die Musik besonders wichtigen Ohreigenschaften berücksichtigen. Diese sind:

1. Die Fähigkeit, die absolute Tonhöhe zu erkennen. Diese Fähigkeit ist besonders groß für hohe Frequenzen (Formanten), die sich periodisch im Rhythmus einer tieferen Frequenz (des Grundtones) wiederholen. Es ist die Klangfarbenempfindung. Wir können Klangfarben unterscheiden, deren Formantfrequenzen um einen Halbton auseinanderliegen. Personen mit absolutem Tonhöhenbewußtsein können die Tonhöhe etwa mit dieser Genauigkeit auch bei sinusförmigen oder schwach oberwelligen Tönen erkennen. Insofern hat die althergebrachte Intervalleinteilung in Halbtöne eine gewisse Berechtigung. Durch Übung von Jugend auf müßten alle Menschen mit normalem Gehör das absolute Tonhöhenbewußtsein erlangen können.

2. Die Fähigkeit der außerordentlich raschen Tonhöhenerkennung. Es genügen in den mittleren Lagen eine, in den höheren Lagen zwei bis drei Perioden, um eine Tonhöhe zu erkennen, so daß hohe Formantfrequenzen schon in ca. 2 ms wahrgenommen werden können, bevor das Ohr den Grundton erfaßt [2]. Das Ohr kann somit in jede Periode des Grundtones ganz unabhängig von dessen Höhe hineinhören. Die alte Vorstellung, welche die Klangfarbenempfindung durch Obertöne und dadurch als abhängig vom Grundton erklärt, ist nicht mehr aufrechtzuerhalten. Sie widersprach von jeher dem phänomenologischen Befund. Durch den Verzicht auf die Obertontheorie der Klangfarbenempfindung verliert auch die Lehre von den Einschwing-, Ausschwing- und Ausgleichsvorgängen ihre Bedeutung. Diese Vorgänge sind spezielle Nuancierungen des Dynamik- und Frequenzverlaufs.

3. Die Fähigkeit, die Frequenzlage von Bandenspektren zu erkennen. Es ist die Geräuschempfindung, insbesondere das Erkennen der Konsonanten der Sprache. Eine Weiterentwicklung der Tonkunst wird dieser Fähigkeit und somit diesem Bedürfnis unseres Gehörs stärker Rechnung tragen müssen.

4. Die Fähigkeit des Tonhöhenvergleichs. Diese ist wesentlich höher als die absolute Tonhöhenerkennung. Sie liegt für Tonfolgen im Bereich größter Ohrempfindlichkeit bei Frequenzänderungen um 3 Promille. Eine Melodik, welche sich auf Halbtonintervalle (6 Prozent) beschränkt, trägt der Wahrnehmungsmöglichkeit und somit dem Wahrnehmungsbedürfnis unseres Gehörs schlecht Rechnung. Eine höher entwickelte Melodik sollte sich von dem Halbtonsystem befreien.

5. Die Harmonieempfindung. Sie entsteht durch die Bildung von (subjektiven) Differenztöne. Wir empfinden gleichzeitige Töne umso harmonischer, auf je kleinere ganze Zahlen im Zähler und Nenner

sich das Schwingungsverhältnis zurückführen läßt. Um so näher liegt nämlich der 1. Differenzton bei ihnen und um so kleiner ist die Anzahl der Differenztöne höherer Ordnung. Unser Gehör kann noch Differenztöne bis etwa zur 7. Ordnung bilden und als Harmonieempfindung wahrnehmen. Bei nicht ganzzahligem Frequenzverhältnis könnte zwar der 1. Differenzton nahe bei den ursprünglichen Tönen liegen, aber es ergäbe sich eine nicht mehr erfaßbare Zahl von Differenztönen höherer Ordnung.

6. Die Lautstärkeempfindung.

7. Das Richtungshören.

8. Die Fähigkeit, aus dem *Lautstärkeverlauf,* dem *Frequenzgang* und den *Richtungen des Raumnachhalls* weitgehende *Schlüsse* auf Größe und Form eines Raumes *zu ziehen.*

Die Gehöreigenschaften führen für die Entwicklung von neuen Musikinstrumenten zu folgenden Forderungen, die nicht alle von einer einzigen Type realisiert werden müssen.

Zu 1. und 2.: Die bisher auch bei elektronischen Musikinstrumenten übliche Methode des Anstoßens von Formantschwingungen durch Unstetigkeiten in der erregenden grundtonhaltigen Schwingung (oder in einer anderen Auffassung das Aussieben von Teiltönen [3]) entspricht nur primitivem Wahrnehmungsbedürfnis unseres Gehörs. Ebenso reizarm wie sinusförmige Töne sind auch sinusförmige Formantschwingungen. Klanglich interessante althergebrachte Instrumente, z.B. die Geigen, sind durch sehr differenzierte Formanten gekennzeichnet. Die Klangfarben der aufschlagenden Zungen, die durch unperiodische Formantschwingungen gekennzeichnet sind, konnten auf elektronischem Wege bisher überhaupt nicht nachgebildet werden (womit nicht gesagt sein soll, daß allgemein die Nachbildung althergebrachter Klangfarben ein erstrebenswertes Ziel elektronischer Klangerzeugung sei!). Abb. 1 zeigt eine Schaltung, welche es ermöglicht, beliebige Formantschwingungen in jede Periode des Grundtons mit gleicher Phase einzublenden. Es ist nur erforderlich, daß die Formantschwingungen mit Hilfe von synchronisierbaren Generatoren erzeugt werden. Den Formantschwingungen kann mit der dargestellten Schaltung jeder beliebige Dämpfungsverlauf erteilt werden. In der unteren Reihe ist angegeben, wie z.B. Formantschwingungen von zunehmender Frequenz erzeugt und periodisch eingeblendet werden können, wodurch Klangfarben ähnlich den aufschlagenden Zungen entstehen.

Die Aufgabe der vielgestaltigen Formantschwingungen kann z.B. auch dadurch gelöst werden, daß die Schwingungsform einer Grundtonperiode auf ein Transparent aufgezeichnet, von dem nach dem Sägezahngesetz pendelnden Strahl eines Kathodenstrahloszillographen durchleuchtet und der Vorgang photoelektrisch in Schall umgesetzt wird.

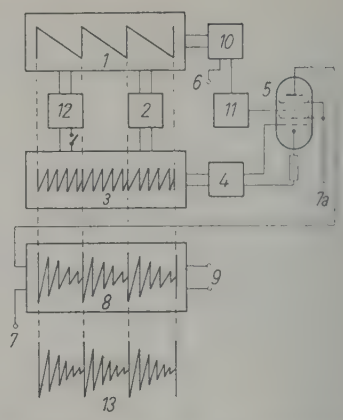


Abb. 1. Klangfarbenbildung durch Formantkippschwingungen.
1. Kippgenerator für den Grundton,
2. Kopplungsglied zu
3. Kippgenerator für die Formantschwingungen,
4. Kopplungsglied zu
5. Amplitudenmodulator zur Erzeugung eines gedämpften Verlaufs,
6, 7, 7a. Speisepunkte,
8. gedämpfte Formantkippschwingungen,
9. Umwandlung in Schall,
10, 11. Vorentzerrer der Amplitudenmodulation,
12. Frequenzmodulation,
13. frequenzmodulierte Formantkippschwingungen.

Auch mit Hilfe von Phonmontagen, wie sie z.B. von MEYER-EPPLER [4] beschrieben werden, sind stärker differenzierte Klangfarben erzeugbar.

Zu 3., 6., 7., und *8.*: Die elektronische Erzeugung von Farbgeräuschen, dynamischen, raumakustischen und stereophonischen Effekten ist bekannt, ihre Einbeziehung in die aktive Tonkunst ist geboten [5].

Zu 4. und 5.: Alle Probleme der Intervalle, der temperierten oder reinen Stimmung, sind nur durch die Unvollkommenheiten der althergebrachten Musikinstrumente entstanden. Nur um auf einem Tasteninstrument spielen zu können, hat man für die Melodik und die Harmonik das gleiche, unserem Gehör wenig entsprechende Intervallsystem gebildet. Es ist an der Zeit, die Musik von diesen Fesseln zu befreien. Melodik und Harmonik sind heterogene Elemente der Tonkunst, sie müssen voneinander gelöst werden. Die Melodik muß so frei wie möglich, die Harmonik so gebunden wie möglich sein. Einer freien Melodik dient ein Manual nach Abb. 2. Die Tasten sind nur Anhaltspunkte. Die Tonhöhenbildung muß dem Willen des Spielers unterstellt sein, jede Mechanisierung ist als unkünstlerisch zu verwerfen. Dagegen ist es nicht sinnvoll, die Harmoniebildung den Musikern zu überlassen. Die Schwingungsverhältnisse müssen den Gehöreigenschaften entsprechend exakt festgelegt werden. Das

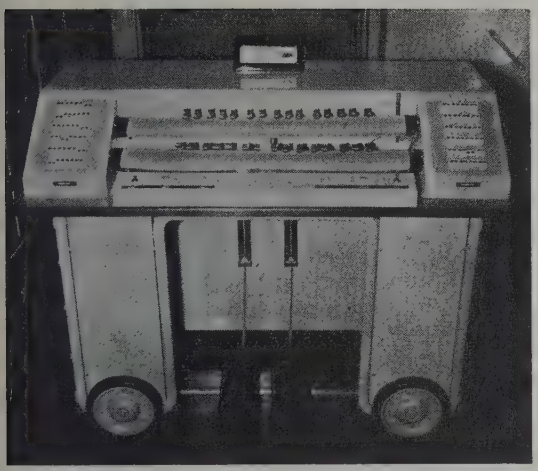


Abb. 2. Das Trautonium des Nordwestdeutschen Rundfunks, Sender Köln. Die Tonhöhe ist eine lineare und kontinuierliche Funktion der Manuallänge, die Lautstärke des Drucks senkrecht auf die Manualfläche. Die Tasten sind Anhaltspunkte zum Auffinden der Intervalle.

besorgt viel besser und richtiger eine zwangsläufig arbeitende technische Einrichtung. Das gegebene Mittel der Elektronik ist die Frequenzdivision von Kippschwingungen. Daß dabei der höchste Ton der führende ist — entgegen der Eigenschaft des Gehörs, sich nach dem tiefsten Ton zu orientieren — ist unwesentlich, es ist dies nur eine Frage der Notierung. Die Obertonreihe wäre für die gestellte Aufgabe höchst ungeeignet, weil die Beibehaltung eines harmonischen Verhältnisses bei einer veränderlichen Frequenz durch die Frequenzdivision technisch viel leichter zu realisieren ist. Das Gehör empfindet die durch die Frequenzdivison erzeugten reinen Harmonien besonders angenehm.

Die elektronischen Mittel und der Bruch mit der Tradition der Instrumente werden die Tonkunst zu einer Evolution führen, in welcher nicht die Enge des Materials, sondern das Gehör das Maß aller Dinge ist.

Schrifttum

[1] Das vom Verfasser entwickelte und nach ihm benannte „Trautonium" ist durch Kompositionen von HINDEMITH, GENZMER, WEISMANN und anderen namhaften Komponisten bekannt geworden und durch das künstlerisch-virtuose Spiel von O. SALA, der das Instrument auch konstruktiv weiterentwickelt hat bis zu seiner neuesten Form, dem „Mixturtrautonium".

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[4] MEYER-EPPLER, W., Melos, Zeitschrift für neue Musik, B. Schott's Söhne, Mainz 1 [1953], 5.

[5] SALA, O., Psycho-physische Konsequenzen elektroakustischer Klangsynthesen. Frequenz 5 [1951], 13.

INHARMONICITY OF PIANO STRINGS *)

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Summary. The natural frequencies of an actual piano string depart somewhat from the harmonic series; the departure is here called inharmonicity. The inharmonicity has been measured for both bass and treble strings in three sizes of Steinway pianos. The bass strings of the usually preferred grand piano have less inharmonicity than have the corresponding strings of a small upright piano. Tuning is correlated with inharmonicity.

Sommaire. Les fréquences propres d'une corde de piano ne correspondent pas exactement à une série harmonique; on appelle ici cet écart l'"inharmonicité". On l'a mesuré pour les cordes de la basse et de l'aigu de trois pianos Steinway de différentes dimensions. Les cordes de la basse du piano à queue, qu'on préfère généralement, présentent moins d'inharmonicité que les cordes correspondantes d'un petit piano droit. L'accord est en relation étroite avec l'inharmonicité.

Zusammenfassung. Die Eigenfrequenzen der Klaviersaite weichen etwas von der harmonischen Lage ab, was als „unharmonisch" bezeichnet werden soll. Dieses wurde sowohl bei Baß- als auch bei Diskantsaiten dreier Größen von Steinway-Pianos gemessen. Die Baß-Saiten des im allgemeinen bevorzugten Flügels haben weniger Abweichungen als die entsprechenden eines kleineren Klavieres. Das Stimmen steht in Zusammenhang mit den unharmonischen Eigenschaften.

1. Introduction

The ideal string is one whose modes of vibration have frequencies which form a harmonic series; that is, the natural frequencies of the string are integral multiples of the fundamental frequency. Actual piano strings are stiff and the frequencies depart more or less from the harmonic series. Such a departure is here called *inharmonicity*. It seems likely

*) Investigation conducted privately.

[1] that the musical quality of a piano and its tuning are related to the inharmonicity.

2. Method of measurement

With the Stroboconn (chromatic stroboscope) [2] sound is picked up by a microphone (or vibration pick-up) and is translated to a flashing light whose frequency is compared stroboscopically with that of a particular pattern on one of twelve rotating disks (one for each chromatic note within an octave). The pattern consists of seven rings having respectively 2, 4, 8, 16, 32, 64, 128 "dark-and-light" segments. The speed ratio of successive disks is very nearly the twelfth root of two, attained by gear ratios 89/84 and 107/101 in alternation. The disks are driven by a synchronous motor whose speed is established by the frequency of an electrically maintained tuning fork which is adjustable over a semitone and calibrated in logarithmic cents. Within an interval of seven octaves any deviation from the equi-tempered scale can be read directly.

A filter such as the General Radio Type 760-A Noise Analyzer or Type 762-B Vibration Analyzer facilitated the selection of the particular frequency desired.

3. Treble strings

The stiffness of the steel wire from which piano strings are made accounts almost entirely for the inharmonicity of the plain wire strings. When one measures and plots the interval in cents by which the frequencies of the different modes of vibration are sharper than the "ideal" harmonic values, a graph is obtained of which Fig. 1 is typical. The open circles represent experimental data on a string whose fundamental frequency is nominally 440 c/s. The slope of the straight line of best fit, shown on the figure, is hereafter called the *coefficient of inharmonicity*.

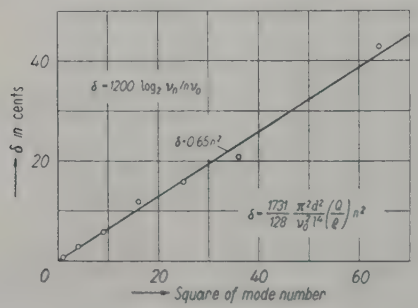


Fig. 1. Plot of inharmonicity δ against square of mode number, for the A_4 string of a Steinway concert grand piano.

The mathematical definition of the inharmonicity δ, in cents, appears at the upper left of Fig. 1; ν_n is the frequency of n-th mode of vibration, and ν_0 is the fundamental frequency of an ideal string computed in the usual way from the length, tension, and linear density. It can be shown [3] that when this defining equation is applied to the classical theory [4] for the vibration of stiff strings the inharmonicity is predicted by the equation at the lower right of Fig. 1. Here d is the diameter of the steel wire, Q is Young's modulus, ϱ is the density of the steel, l is the length of the string. When the dimensions of the particular A_4 string are substituted in this stiffness equation the coefficient of inharmonicity turns out to be exactly that found empirically: i.e., 0.65 cent/n².

It has been demonstrated [3] that this equation describes very well the observed inharmonicity of the plain wire strings in six different pianos. The computed coefficients for three of these pianos are depicted at the right of Fig. 2. A line is simply a guide; the coefficient really has a discrete value for each particular string which may be identified by the note name and subscript; by the notation [5] employed middle C is C_4. It is evident that the inharmonicity increases regularly as the strings become shorter. Gaps in the curves direct attention to discontinuities associated with the presence of bars in the iron frames. Such a discontinuity does not always, however, occur at a bar.

4. Bass strings

When one plots the inharmonicity of bass strings in the manner of Fig. 1, a reasonably straight line fits the data above the fourth mode of vibration, but for the lower-numbered modes there are frequently noticeable deviations. The inharmonicity for the first mode of vibration is often several cents below the extrapolated straight-line trend.

From the straight portions of the inharmonicity plots a coefficient of inharmonicity was estimated visually for each bass string in the three pianos here under consideration. The plotted points in Fig. 2 represent these coefficients. As an aid in judging the significance of a coefficient of 0.5 cent/n², for example, note that under such circumstances the fifteenth mode of vibration has a frequency more than sixteen times that of the fundamental frequency.

Lines fitted visually to these bass string coefficients are also given in the figure. The gap between $F\sharp_1$ and G_1 in the Style 40 curve and the gap between E_1 and F_1 in the Style A curve call attention to the pronounced changes in trend at these positions; strings to the left are single while those to the right are double. Single, double, and triple bass strings

are used in the Style D piano but no marked changes in the coefficient occur for this piano.

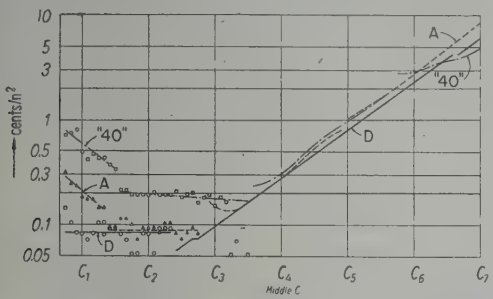


Fig. 2. Coefficient of inharmonicity for Steinway concert grand piano (Style D), 6-ft grand (Style A), and upright (Style 40). Experimental coefficients shown for bass strings only.

5. Some musical implications

Although pianos are nominally tuned in equal temperament, measurements on a variety of pianos show that the treble strings are tuned consistently and progressively sharp: the strings in the top octave may be relatively more than 20 cents sharp. If one postulates that in tuning by octaves the fundamental frequency of the upper string is made equal to the frequency of the second mode of vibration of the lower string, then the cumulative effect is a tuning somewhat like that observed. Strings in the extreme treble, however, are apt to be tuned erratically and even sharper than the computed trend.

The fundamental frequencies of bass strings, on the other hand, are tuned consistently flat; the deviations from exact equal temperament are slight in a large grand piano but are very appreciable in smaller pianos. Thus, at least qualitatively, the deviations in tuning are greater for these strings having greater inharmonicity. For example, the C_1 string of a Style 40 upright piano was tuned 50 cents flat, whereas the same tuner tuned the corresponding string on the Style D piano only 8 cents flat.

In discussions of tone quality it is customary to rely upon the classic definition [6] that ". . . differences in musical quality of tone depend solely on the presence and strength of partial tones . . ." While HELMHOLTZ recognized that the tone of some instruments is dependent upon the way it begins and ends, he chose to restrict his attention to the "peculiarities of the musical tone which continues uniformly" [7] and to consider as musical tones only those with harmonic upper partials [8].

The sound of a piano string, however, is not steady and the partials are inharmonic. Also the bass strings of the usually preferred large grand piano have less inharmonicity than have the corresponding strings of the small upright piano. More attention should be devoted to the transient parts of musical sounds and in particular the frequency of the components. It appears that a significant description of a musical sound must include not only the amplitudes (and decay rates) of the partials but also their precise frequencies.

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Discussion

SKUDRZYK, E.: The curves shown seem to give very interesting cues on the mechanism of hearing at high and low frequencies. Has any work been done in this respect?

LECTURER: I am not aware of any direct study of inharmonicity in relation to hearing. It would be valuable to establish what inharmonicity is just detectable by the ear and how musical quality is judged to change with inharmonicity.

SKUDRZYK, E.: In case of stationary tones (forced oscillations) the overtones would be expected to form a harmonic series, as the process is periodic. In case of the piano the sounds are transcient and the overtones not necessarily harmonic. This seems to make a great difference in the sounds.

LECTURER: I agree. It would be interesting to compare the sound of the usual piano with one in which the strings would produce harmonic partials.

MOLES, A.: An experiment in which a score being transposed to the lower octave and played rather slowly by an artist, recorded and played back at double speed to bring back the sounds to their original range, gives a queer tone even with a well tuned pianoforte. Do you think that this particular effect may be explained in terms of inharmonicity?

LECTURER: Yes, I think it very likely that the resulting tone can be explained in part as due to the different inharmonicity in the strings an octave apart.

THIENHAUS, E.: 1. Unharmonische Teiltöne sind qualitativ leicht am Kathodenstrahloszillographen zu beobachten (Auswandern der kleineren Zacken bei Synchronisierung des Grundtones), 2. Die unharmonischen Teiltöne müssen als wesentlich für die musikalische Tonqualität angesehen werden, da dadurch die für die Musik unerläßliche Lebendigkeit entsteht, während Klänge mit exakt harmonischen Teiltönen einförmig und langweilig klingen. Es wäre interessant, ähnliche Untersuchungen auch an Blasinstrumenten durchzuführen. Grundsätzlich hat man bei allen Schwingungserzeugern (Generatoren, Saiten, Luftsäulen), die *nicht* in strengem Sinne eindimensional sind, unharmonische Teiltöne zu erwarten.

LECTURER: In accordance with the remark just made by Prof. SKUDRZYK, when a wind instrument is maintained in steady vibration the resulting waveform observed on a cathode ray oscilloscope is periodic, so that the partials making up the complex sound must be harmonic. However, the frequencies of the different modes of vibration produced by the player (as when he plays different notes with the same fingering) are *not* in general harmonic (see brief description in Amer. J. Phys. 20 [1952], 177 ··· 183). It is to be expected, therefore, that even for musical instruments capable of producing steady tones the transient part of their sound consists of inharmonic partials. It has been long recognized that many instruments can be distinguished principally by the transient and it may well be that the ear is particularly sensitive to the inharmonic relationships of the partials which make up the transient.



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SECTION VII

SYMPOSIUM ON THE SOUND INSULATION OF LIGHTWEIGHT STRUCTURES

GENERAL REVIEW

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Laboratory for Technical Physics, Technological University, Delft



1. Introduction

There is a tendency in modern building practice to eliminate all material which is not absolutely necessary from the viewpoint of stability or thermal insulation. Movable partitions in offices, laboratories and so forth are examples. However, the requirements for insulation against airborne sound tend to become more stringent as a consequence of the increasing intensity of life. These factors, seen against the background of the mass-law of sound insulation — which states that the mass per unit area or the surface density of a wall is the governing factor for insulation —, form the problem with which architectural acousticians are confronted. How can we achieve sufficient insulation with less material; or in other words, how can we violate the mass law?

As variables in this process of improvement we have at our disposal: the flexural stiffness, the internal damping and the damping at the boundaries if simple single walls are concerned, while with cavity walls we have in addition: the distribution of the total mass over the two or more leaves of the construction, the thickness of the cavity, the possible filling material etc. Enemies of good insulation are sound-bridges in cavity walls, leakage of sound through slits between the elements constituting the construction, flanking transmission, etc.

The study of lightweight structures is tempting for various reasons. The insulation of such structures is easy to measure in the laboratory as well as in the field without the disturbing effect of flanking transmission. Boundary conditions in the laboratory can rather easily be adapted to those assumed by theory, thus facilitating theoretical discussion of the measurements. Many lightweight structures show clearly the influence of flexural stiffness, which may lead to the complete understanding of the influence of flexural stiffness. Theories concerning lightweight structures are in rapid development; they require discussion and experimental verification.

It therefore seems justifiable to expect that the study of the sound insulation of lightweight structures will contribute considerably to the whole understanding of airborne sound insulation, including that provided by heavy walls.

I. SINGLE WALLS

2. The mass controlled incompressible single wall

If the wall is considered to have a negligible flexural stiffness and also to be incompressible, its surface density (in kg/m^2) must govern the insulation. For sound at normal incidence we may use a formula due to RAYLEIGH [1]; in modern symbols it reads (Fig. 1, curve 1):

$$i_0 = 10 \log [1 + (\omega m/2\varrho c)^2] \qquad (1)$$
$$\simeq 20 \log \omega m/2\varrho c$$

where

i_0 = insulation or transmission loss at normal incidence

= sound pressure level difference between incident and transmitted wave in dB;
ω = $2\pi \times$ frequency f;
m = surface density of the wall in kg/m²;
ϱc = specific acoustic impedance of air ≃ 420 MKS-rayl *), ϱ being the density of air (1.2 kg/m³) and c the velocity of sound in air (344 m/s).

Eq. (1) holds also for walls of finite dimensions, flexural stiffness still being supposed excluded.

The insulation i_θ at an oblique angle of incidence θ is smaller [3], namely

$$i_\theta = 10 \log [1 + (\omega m \cos \theta / 2 \varrho c)^2]. \qquad (2)$$

The derivation of eq. (2) follows the same simple lines as that of eq. (1), the factor $\cos \theta$ originating from the fact that at oblique incidence the normal components of the air velocities enter instead of the total velocities, and the former are $\cos \theta$ times smaller.

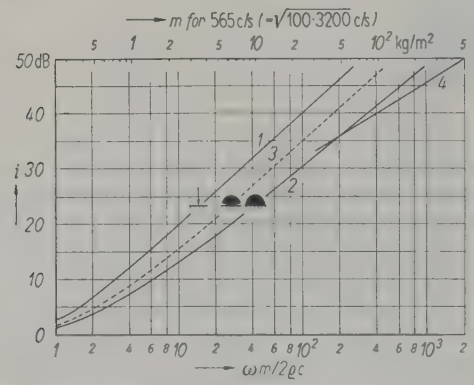


Fig. 1. Mass laws;
curve 1: normal incidence,
curve 2: random incidence,
curve 3: ditto restricted to θ = ca. 80°.
For comparison: curve 4 = DIN 4109 = mean insulation in practice at m-values of the upper scale.

By suitably averaging eq. (2) over the angle of incidence θ, the so called "mean insulation" for random incidence is obtained. The result [4] (Fig. 1, curve 2) is

$$i_m = i_0 - 10 \log 0.23\, i_0. \qquad (3)$$

The mean insulation i_m is therefore smaller than the insulation at normal incidence by the amount 10 log 0.23 i_0 dB, a "correction" which increases with i_0 and exceeds 10 dB for $i_0 > 44$ dB.

The major contribution to the correction is due, however, to transmission at almost grazing incidence. CREMER [4] argues that it seems more reasonable to limit the angles of incidence to $\theta = 75°$ or 80°. The correction then becomes equal to 5 dB independent of i_0, with the result that

$$i_m \simeq i_0 - 5 \text{ dB}. \qquad (4)$$

It can hardly be expected that the simple theories reported here would agree with experiment. Indeed there are systematic discrepancies. Roughly speaking, we can say that the insulation at high frequencies is smaller than that predicted by mass law, whereas the reverse is often true at very low frequencies. On the average the insulation between 100 and 3200 c/s is generally lower than that predicted by mass law. Moreover, as a result of the two effects mentioned, the average slope of the insulation-frequency characteristic is smaller than 6 dB/octave, say 4 to 5 dB/octave only.

The high-frequency loss in insulation has been discerned and explained by CREMER [4]; it originates in the flexural stiffness. The excess insulation at low frequencies was pointed out by LONDON [5]; mathematically it could be accounted for by the introduction of a resistance, but from a physical point of view it is not yet understood.

Fig. 1, curves 1, 2 and 3, shows the theoretical predictions for the various cases. The insulation i is plotted against the variable $\omega m / 2 \varrho c$. It has been tentatively standardized that in cases where an average figure is required as a quality index for a given wall, the mean dB-value is to be computed by averaging i between 100 and 3200 c/s on a logarithmic frequency scale. In view of the approximately linear theoretical relation between i and log ω, this is roughly equivalent to taking the insulation at the geometric mean frequency between 100 and 3200 c/s, i.e. 565 c/s. For the sake of convenience we have therefore added a second abscissa scale (top of Fig. 1) referring to that frequency.

For comparison we have also included the German DIN-standard curve 4 relating the surface density m (upper scale) to the mean insulation found in practice.

3. The influence of flexural stiffness

The general case of walls having mass, flexural stiffness, damping and finite dimensions and excited at oblique incidence cannot yet be adequately handled mathematically. Confining attention to normal incidence SCHOCH [6] has shown that the mass law holds asymptotically for sufficiently high frequencies. CREMER [4], ignoring the restriction to finite dimensions, finds that for infinite walls the mass law holds asymptotically for *low* frequencies, irrespective of the angle of incidence.

*) The name rayl (from RAYLEIGH) was suggested by BERANEK [2], unfortunately however for the CGS-unit, which may be assumed to disappear together with the CGS-system within the next 10 years. The reviewer suggests that in the meantime we should distinguish between the MKS-rayl and the CGS-rayl, thus keeping RAYLEIGH's name for this unit.

SCHOCH's reasoning, reproduced here in a few words, amounts to saying that at normal incidence only the lower modes of vibration of the finite wall are considerably excited and these will be mass-controlled at sufficiently high frequencies, i.e. well above their corresponding resonance frequencies.

CREMER [4] studies the forced flexural wave which the incident sound imposes on the infinite wall at oblique incidence. Its velocity is dictated by the velocity of sound in air c and the angle of incidence θ, thus

$$c_{forced} = c/\sin\theta. \tag{5}$$

A free flexural wave, which must be clearly distinguished from the forced wave just mentioned, may also travel through the wall. Its velocity turns out to be frequency-dependent, thus

$$c_{free} = \sqrt[4]{\frac{Ed^3}{12\,m}} \cdot \sqrt{\omega}, \tag{6}$$

where E is the modulus of elasticity in N/m^2 and d the thickness of the wall in m.

Now if the forced velocity c_{forced} becomes equal to the free velocity c_{free}, the wall has lost every resistance against transmission. A "free" wave will be "forced", having such an amplitude that total transmission occurs. Bearing in mind (see eq. (5)) that c_{forced} is always greater than c, but for grazing incidence ($\theta = 90°$) when c_{forced} equals c, it is readily seen from eq. (6) that for fulfilling the coincidence condition $c_{forced} = c_{free}$ the frequency must exceed a critical frequency determined by

$$c_{free} = c$$

yielding (Fig. 2)

$$f_{crit} = \frac{c^2}{2\pi d}\sqrt{\frac{12\,\varrho_w}{E}}, \tag{7}$$

ϱ_w being the density of the wall. Above this frequency $c_{free} > c$, and there can always be found an angle of incidence for which

$$c_{forced} \equiv c/\sin\theta = c_{free},$$

i.e. for which total transmission occurs. Thus total transmission corresponds to coincidence of the forced and free waves. It takes place for grazing incidence at the critical frequency and at angles $\theta < 90°$ at higher frequencies.

At frequencies well below f_{crit} the elements of the wall behave independently of each other, i.e. the wall behaves as mass-controlled. The elements are coupled by elastic forces, but the vibration of an element of the wall cannot be influenced by the vibration amplitudes of the neighbouring elements, since the velocity with which the information between the elements travels along the wall, i.e. c_{free}, is too small.

The case of total transmission is of course idealized. If the wall loses its mass character by stiffness interaction, it does not become quite resistanceless, but simply resistance-controlled.

Thus at a given angle of incidence and increasing frequency the wall is mass-controlled at low frequencies, resistance-controlled near coincidence and by extrapolation it may be inferred that the system becomes stiffness-controlled at frequencies higher than the coincidence frequency. So in a certain sense the behaviour is just the opposite to that with resonance. That coincidence has nothing to do with ordinary resonance is evident from the fact that resonance in wave problems originates from boundary conditions, whereas for the time being the wall is supposed to be indefinitely extended.

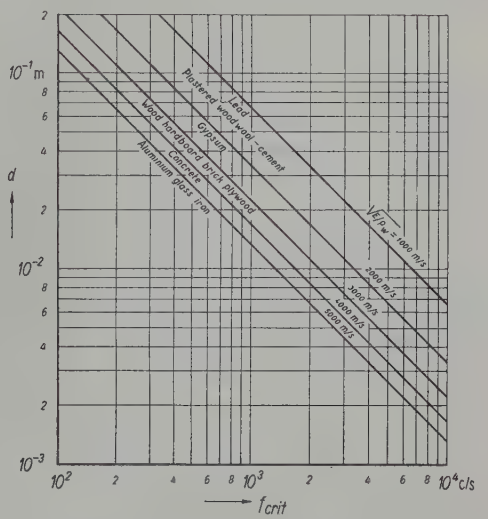


Fig. 2. The critical coincidence frequency as a function of wall thickness (d) and longitudinal sound velocity $\sqrt{E/\varrho_w}$ in the wall [4].

Another essential difference between coincidence and resonance is that the critical coincidence frequency decreases with the wall thickness d (Fig. 2), whereas the resonance frequencies of finite plates increase with d.

We quote the following formula directly from CREMER's article:

$$i_\theta = 10\log\left\{1 + \left[\frac{\omega m\cos\theta}{2\,\varrho\,c}\left(1 - \frac{c^4_{free}\sin^4\theta}{c^4}\right)\right]^2\right\} \tag{8}$$

the quantity between ( ) being the "correction" due to stiffness.

CREMER has moreover calculated the insulation for random incidence with and without inclusion of internal damping in the wall material. As expected, a breakdown in the mass law curve occurs at the critical frequency.

Numerous experimental results corroborate the theory. One single verification is shown in Fig. 3 [7]. Rubber is not expected to suffer much from

coincidence, since it is flexible and must have a high critical frequency. It closely obeys the mass law. The equally heavy steel and concrete slabs clearly show a considerable coincidence breakdown at the frequencies predicted by theory.

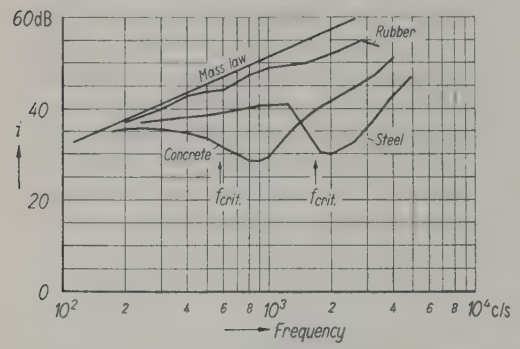


Fig. 3. The random incidence insulation after GÖSELE [7] of four walls of 55 kg/m².

Coincidence breakdown in the middle of the frequency range, say between 500 and 1000 c/s, should in any case be avoided. Walls having their coincidence at, say 2000 c/s (lightweight structures) should be made more flexible in order to shift the critical frequency to still higher frequencies [8] whereas the reverse applies to walls with critical frequencies around 200 c/s.

SCHOCH's result [6] was too complicated for use at any but normal incidence, CREMER's [4] was confined to the case of infinite walls. To cover the practical case, a synthesis of these points of view has recently been considered both experimentally and theoretically by SCHOCH and FEHÉR [9]. Their starting point is CREMER's result. The forced flexural wave in a finite wall is incompatable with the boundary conditions without the introduction of auxiliary free waves travelling from the boundaries inwards. Using a simplified picture we could say that the forced wave, on arrival at the boundary, is reflected as a free wave, since it has lost its guide, the air wave. The free wavelength will be smaller than the wavelength in air for frequencies below the critical frequency. Such short waves radiate very weakly into the surrounding medium [10], from which it is evident that the mass law should apply reasonably well to finite walls below the critical frequency. Above this frequency the free waves which are reflected do radiate. These free waves, on meeting each other on their way inwards, may give rise to resonance, and thus lead to increased radiation, i.e. to transmission. On the average, we should expect that these free waves from the boundaries would lead to an increased transmission as compared to that of the infinite wall. Striking and instructive observations of this effect measured on a model scale are to be found in [9].

4. Insulation at low frequencies

As mentioned above, the insulation at low frequencies — well below the coincidence breakdown — is frequently better than that predicted by mass law [5], [9]. Similar unmistakable results have recently been obtained by PEUTZ in the reviewer's acoustic department. LONDON, who laid considerable stress on this effect, accounted for it by the introduction of a resistance (of viscous or other origin) contributing to the mass reactance. As he has shown, this resistance is essential to enable double walls with unfilled cavity to insulate better than single walls with the same overall surface density. This resistance, therefore, is an important feature of a single wall. Its nature is as yet not understood. It has been suggested [9] that there is something wrong at low frequencies with the method of correction for absorption in the receiving room.

II. DOUBLE WALLS

5. The mass-controlled double wall with unfilled cavity

The theory of transmission at normal incidence for this case is well known [11]. The insulation curve shows a resonance dip at a generally rather low frequency of say 100 c/s, at which frequency the two leaves vibrate in opposite phase against the air spring constituted by the air in the cavity. Then we should expect a steep increase in insulation, leading to values of the insulation considerably above those predicted by mass law for the equally heavy single wall and extending up to the frequency range of, say, 2000 c/s and higher, at which point the insulation begins to suffer from resonances of the air itself within the cavity. On the average this theory for normal incidence predicts a considerable superiority of cavity walls over equally heavy single walls. The fact that practical cavity walls afford generally poorer insulation than this theory would predict is usually accounted for by the use of too heavy wall ties and other sound bridges.

LONDON [12] has shown however that at random incidence a cavity wall with unfilled cavity *does not constitute an improvement* over the equally heavy single wall. Above the rather low resonance frequency f_0, the double wall is as a rule even inferior in comparison with the equally heavy single wall. This is due to the fact that the resonance dip in the curve shifts to higher frequencies for oblique incidence in the ratio $1 : \cos\theta$. It is readily seen, however, that a shift of the resonance to, say $5f_0$ requires almost grazing incidence. It seems doubtful

therefore whether the low theoretical insulation predicted at frequencies above $5f_0$ will actually appear in practice. We would suggest instead that the resonance phenomenon at random incidence reduces to a breakdown over a *limited* frequency range of say a few octaves, as in the case of the coincidence effect.

For structures of very light weight this may be serious enough, however, since their f_0 is between, say, 100 and 300 c/s so that their resonance dip for random incidence falls in the important frequency range. Something should be done, therefore, to counteract the resonance of lightweight structures occurring at oblique incidence. Possible means would be to increase the internal damping both of the wall material and at the boundaries, and to use small quantities of sound-absorbing material in the cavity near the boundaries, or preferably throughout the whole cavity.

LONDON assumed that his unexplained but necessary resistance, introduced to account for the observed insulation of single walls especially at low frequencies, would also be present in both leaves of a cavity wall. LONDON [12] then calculated that the random insulation of a cavity wall would be superior to that of an equally heavy single wall by some 10 to 20 dB, and found agreement between theory and experiment on this point. This provides another demonstration that LONDON's resistance must have a "raison d'être".

6. The unfilled double wall with flexural stiffness

It is clear that a coincidence dip of one of the leaves impairs the overall insulation of the cavity wall. One of the leaves is, then, so to speak ineffective in the frequency range involved. If both leaves have their coincidence dip in the same range the damage to the curve may be serious indeed. It therefore seems wise to choose the leaves of different thickness and/or of different material in order to avoid overlap of the coincidence dips of the leaves. This is a well-known rule of thumb probably based only on the architect's instinct or feeling. It is often "explained" by emphasizing that the resonance frequencies of both leaves would then be different, but in our opinion this reasoning is theoretically unsound. As a rule really lightweight structures can easily be manufactured so flexible that the coincidence dips lie at rather high frequencies so that it does not seem necessary to pursue the matter further. It should be emphasized, however, that it is important to verify that the critical coincidences are really sufficiently high.

An instructive result due to FLEMING [13] is reproduced in Fig. 4 (see legend). It can be explained by the following reasoning.

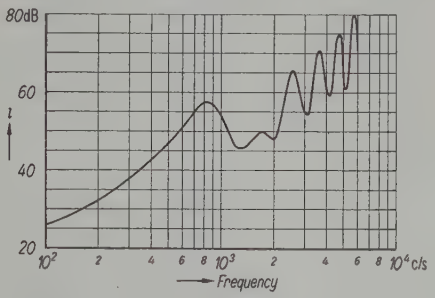


Fig. 4. Random incidence insulation of double window after FLEMING [13], leaves of 8 and 9.5 mm respectively, 15 cm cavity (measured points not shown).

If white noise were directed at random incidence onto a single wall, the transmitted sound with frequencies above the critical coincidence frequency would be directionally coloured, for the higher frequencies would prefer the "more perpendicular" transmission, and the lower the more oblique to grazing transmission ($c_{free} = c/\sin\theta$), i.e. the sound above the critical frequency would be more or less spread out into a continuous spectrum, just as would a light beam passing through a prism. If two such leaves are combined to constitute a cavity wall, the continuous dispersed spectrum of the first leaf is transmitted into the cavity. Certain rays will fit into the cavity as to direction and frequency; in acoustical terms they are in resonance, or in optical terms they interfere. These rays therefore excite the second leaf more strongly than do the others. The second leaf has no appreciable resistance against these particular rays, since it too is in coincidence for all of them. The high-frequency part of a random white incident sound is therefore transmitted as a directionally coloured line spectrum. The line spectrum above 2000 c/s in Fig. 4 could be explained quantitatively by this reasoning [13].

Of course a cavity wall with more unequally thick leaves would also show coincidence effects, but they would be less pronounced since the number of dips would be doubled, while their band width and depth would be smaller.

7. Advantages of filling the cavity

It can be shown that the low resonance dip in the transmission curve at normal incidence, caused by the two leaves vibrating against the air spring in between, can be filled out effectively by using a porous flexible filler in the cavity [14]. This might be the case a fortiori at oblique incidence thus doing away with the whole low frequency resonance

phenomenon. In the middle frequency range, say 300 ··· 1000 c/s, the effect of the filler is small for cavities of practical dimensions. At the higher frequencies, however, the insulation is greatly improved. The effect is the less important the thinner the layer of air and the heavier the leaves [12]. Indeed for cavities of a few centimeters the effect is practically restricted to the avoidance of the low-frequency resonance.

The problem of finding the best filler has not yet been completely solved. It is probable that the best material should have a rather high air resistance and a rather high stiffness, though smaller than that of air. Moreover a high internal damping of the filling material is beneficial.

8. The effect of wall ties, common clamping an other sound bridges

This problem is still in need of further study, and will be dealt with during the Congress by several authors. We therefore confine ourselves to a few remarks.

The more flexible the leaves the less dangerous are the sound bridges. A sound bridge excites free waves in the second leaf. In very flexible leaves these waves will have a very short wavelength, and such short waves radiate very inefficiently. This statement, made from the standpoint of theory, is corroborated by experiment [15].

III. SOME INTERESTING MEASUREMENTS

In Fig. 5 LONDON's results [11] on plywood double walls are reproduced, showing (1) the detrimental effect of coincidence near 2000 c/s and (2) the advantage of using a rather wide cavity. The improvement over the single leaf is more than 6 dB and could be explained only by the introduction of a resistance term, which is itself still in need of explanation.

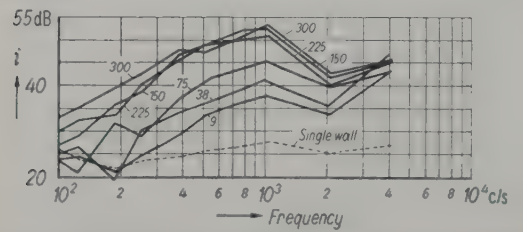


Fig. 5. Plywood double walls with various cavities at random incidence after LONDON [12] as compared with the one-leaf wall ($\frac{1}{2}$ mass). The thickness of the cavity in mm is indicated along the curves.

In Fig. 6 LONDON's experiments [12] on plasterboard double walls are shown; these are analogous to those of Fig. 5.

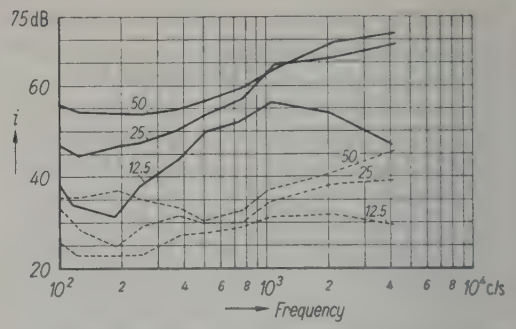


Fig. 6. Plasterboard single and double walls with 7.5 cm cavity; thickness of leaves indicated along the curves in mm (after LONDON [12]).

Fig. 7 shows some results of GÖSELE [16] on a hardboard double wall with fibre slabs in a 15 cm cavity without sound bridges. The improvement is considerably better than 6 dB. The steep decrease near 3000 c/s agrees with the calculated coincidence dip.

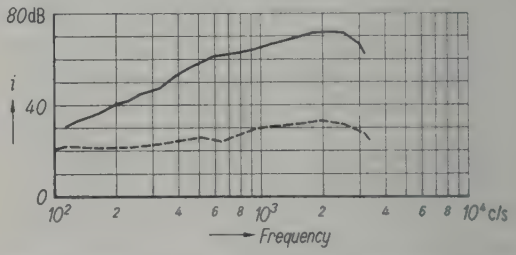


Fig. 7. Double wall: leaves of 1.1 cm hardboard, 15 cm cavity with fibrous filler, no sound bridges or flanking transmission. For comparison the one-leaf wall (1.1 cm) (after GÖSELE [16]).

Fig. 8 (also from GÖSELE [16]) shows clearly the coincidence dips in the single and double wall curves, as well as the resonance dip, calculated at 200 c/s, due to the use of a too shallow cavity. The studs connecting the two leaves every 56 cm seem tolerable. This can be explained below the coincidence dip by the high flexibility of the leaves; for the free waves excited in the second leaf by the

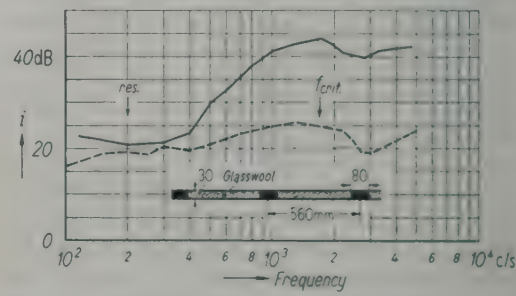


Fig. 8. Double wall with studs: leaves 1 cm wood, 3 cm cavity with glasswool. For comparison: single wall 1 cm wood (after GÖSELE [16]).

studs are of such short wavelength that they would radiate inefficiently. Above coincidence another explanation must be sought. LONDON's resistance of the leaves as well as their additional damping by the glasswool filler may be involved here.

Finally, we show in Figs. 9 and 10 two graphs, containing as yet unpublished results of the Technical Physics Department TNO and TH at Delft. The walls tested had an overall size of 1.9 × 1.9 m² and consisted of a wooden frame F (7.5 × 3 cm²) clamped tightly into the measuring opening between two reverberant rooms in the laboratory. Two prefabricated units of 0.9 × 1.84 m², each consisting of a wooden frame of 7.5 × 3.5 cm² to which the two leaves of the double wall were screwed at both sides, were inserted into the wooden frame F with a stud in between them, the latter being connected with F.

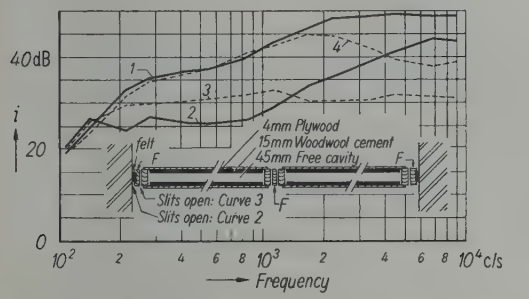


Fig. 9. Double wall as measured in the test house, without leakage of sound through slits (curve 1) and with slits of various kinds (curves 2, 3, 4) (unpublished results Techn. Phys. Dept. Delft).

Fig. 9 shows the detrimental effect of leakage of sound through slits, all curves relating to the same double wall consisting of leaves of 0.4 cm plywood to which 1.5 cm woodwool-cement slabs were stuck at the cavity side, the free cavity being 4.5 cm. Curve 1 shows the insulation of the wall when all slits had been carefully closed with plasticine; curve 2 the insulation with the slits between F and the test house opened, and curve 3 the insulation with the slits between F and the units opened; curve 4 relates to an only partially successful trial to suppress the latter transmission with rubber strips over the slits between F and the units.

The insulation loss due to the slits is in quite good agreement with theory. Slits of approximately 10 cm depth resonate at 1500 c/s (curve 3). The slits between F and the test house (curve 2) resonate at an even lower frequency as a consequence of the (felt-filled) cavity within the stud.

Fig. 10 shows the insulation of four demountable walls. Their description, surface density and mean insulation value are given in the legend of the figure.

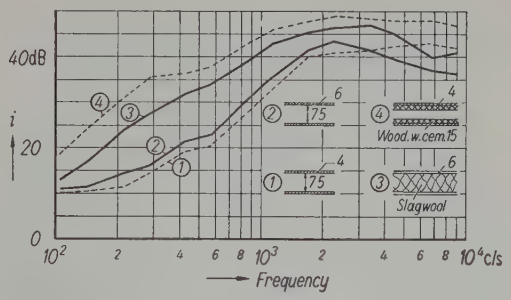


Fig. 10. Double walls consisting of two units 1 × 2 m², analogous to those shown in Fig. 9, measured in the laboratory. Studs of 75 mm; m = total surface density excluded studs in kg/m².

Curve	Construction of a leaf	Cavity	m	mean i
1	4 mm plywood	empty	4.5	24
2	6 mm plywood	empty	6.8	26
3	6 mm plywood	tin slagwool	19.5	33
4	4 mm plyw.+ 15mm woodwool cement stuck to the plywood at inside	empty	21.5	37.5

The improvement of 7 dB of curve 3 over curve 2 is due to the slagwool filler, the 13.5 dB difference between curves 4 and 1 comes from the 15 mm woodwool cement plates stuck to the plywood at the cavity side. Another interesting statement can be made on comparing construction 4 with an 11 cm brick wall. The latter is 10 times heavier and isolates only 4.5 dB better (42 dB) instead of some 14 to 20 dB as predicted by the slope of the various mass laws.

Our conclusions may be summarized as follows:

1) Single walls roughly follow the mass law at low frequencies, but show a coincidence dip at high frequencies.

2) At very low frequencies the rather high insulation of single walls requires some sort of resistance for its explanation.

3) Double walls without losses are theoretically expected to be inferior to equally heavy single walls, at least below the coincidence region and probably also within it.

4) Practical double walls are undoubtedly superior to single walls, the more so the bigger the cavity, the higher the coincidence frequency of the leaves and the greater the damping.

5) Fillers in the cavity are advantageous, especially for very light walls.

6) Sound bridges, although not beneficial, are tolerable, certainly if the leaves are flexible enough.

7) Aspects needing further study are:
   a. the influence of filling materials;
   b. the influence of sound bridges;
   c. the influence of damping;
   d. LONDON's resistance.

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SOME MEASUREMENTS ON LIGHTWEIGHT DOUBLE WALLS

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Summary. Some results of measurements on lightweight double walls are given to show the influence of wall mass, airspace and friction. When introducing heavy sound bridges between two lightweight panels the low-frequency insulation may be improved.

Sommaire. On donne quelques résultats de mesures faites sur des parois doubles de faible poids, pour montrer l'influence de la masse de la paroi, de l'intervalle d'air et du frottement. On peut améliorer l'isolement acoustique aux basses fréquences en introduisant des ponts acoustiques lourds entre deux panneaux légers.

Zusammenfassung. Einige Ergebnisse von Messungen an leichten Doppelwänden werden angeführt, um den Einfluß von Wandmasse, Luftpolstertiefe und Reibung zu zeigen. Durch Anbringung von schweren Schallbrücken zwischen zwei leichten Platten kann die Isolation für tiefe Frequenzen verbessert werden.

1. Introduction

As was first shown by CREMER [1] it is important to decrease the stiffness of a single lightweight wall in order to remove the usual dip in the insulation curve — due to the coincidence effect — to frequencies above the range 100 ··· 3200 c/s. LONDON [2] has demonstrated that some considerable improvement may be gained by increasing the friction of a light panel. When high insulation is aimed at and a low weight is of importance only double or multipanel walls can be accepted. In the following some results of measurements on double walls are given to demonstrate the effect of some wall constants. The measurements have been carried out according to the proposed international standard methods.

2. Double walls without common ties

Theoretical considerations have led to closer and closer agreement between calculated and measured reduction factors of double panels, but it is still not possible to predict the reduction factor with sufficient accuracy. WINTERGERST [3] was the first one to show the trends for double walls without common ties. He however restricted himself to normal incidence and assumed the panels to have mass reactance only. CREMER's studies of the stiffness of a single panel exposed to sound waves at oblique incidence enabled LONDON [4] to take mass, stiffness and friction of a double wall into consideration. LONDON calculated the ratio A of the pressure p_i of the wave falling at the angle θ on the first panel and the pressure p_t of the wave leaving the second panel:

$$A = \frac{p_i}{p_t} \qquad (1)$$

$$= 1 + \frac{Z_w \cos\theta}{\varrho c} + \left(\frac{Z_w \cos\theta}{2\varrho c}\right)^2 \cdot (1 - e^{-j\,2kd\cos\theta})$$

where:

Z_w = mechanical impedance per unit area of each of the two panels,
d = distance between the panels, in the following called airspace,
$k = \omega/c = 2\pi/\lambda$ = wavenumber,
ϱc = characteristic impedance of air.

The corresponding expression for a single panel is:

$$a = \frac{p_i}{p_t} = 1 + \frac{Z_w' \cos\theta}{2\,\varrho\,c}. \qquad (2)$$

Eq. (2) may be derived from (1) by setting $d = 0$ and $Z_w' = 2\,Z_w$.

It is seen that A and hereby the reduction factor in the case of double walls depends to a much higher degree on the wall impedance than in the case of single panels. Consequently, it is more important to avoid the coincidence effect for the double panel than it is for the single panel, i.e. the stiffness should be reduced as much as possible. As the impedance increases with frequency the improvement of the double wall compared to the single wall does as well. LONDON has modified CREMER's expression for the wall impedance by taking the friction R into consideration:

$$Z_w = \mathrm{j}\,\omega m\left(1 - \frac{f^2}{f_c^2}\sin^4\theta\right) + \frac{2\,R}{\cos\theta} \qquad (3)$$

m = panel mass per unit area
f_c = critical frequency of coincidence.

In order to improve the impedance the mass, the friction and the coincidence frequency f_c should be increased, the latter by decreasing the stiffness.

The influence of the airspace is seen by developing the exponential in eq. (1), which reveals that a big airspace is necessary in order to insure a high reduction factor for a light double panel at low frequencies.

Some uncertainty as to wall friction, edge conditions and angular distribution of the waves falling on the panel makes it extremely difficult to calculate the reduction factor from primary constants but the formulae are useful indeed in showing the trends if the wall constants are changed.

For a double wall the important constants to be decided upon when choosing a wall are the mass per unit area of both panels and the airspace. In Fig. 1 is shown the reduction factor averaged within the frequency range of 100 ··· 32000 c/s as a function of airspace for a light and a relatively heavy double wall. The slope of the insulation curves — all similar in shape — falls off gradually as the low frequency insulation is raised due to the increased airspace.

Only moderate improvement over the equally heavy single wall is obtained when the airspace between two light panels is small. LONDON has shown that the improvement may even be negative

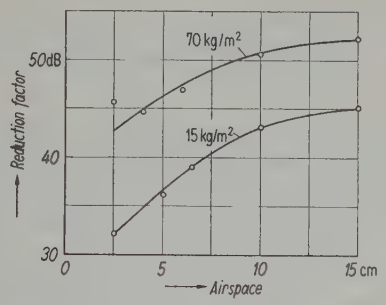


Fig. 1. Reduction factor averaged within 100 ··· 3200 c/s for 70 and 15 kg/m² panels without sound bridges and mounted in separated brick walls. Rockwool blanket in the airspace. Panel dimensions: 2 × 2 m².

at low frequencies when d and m are small and the wall impedance is only due to mass reactance. That this may actually be the case is shown in Fig. 2 where a double plasterboard wall is seen to have a lower reduction factor at frequencies below 400 c/s than the single panel; the latter even had only the mass of *one* of the leaves of the double wall. This effect disappears for bigger airspace and does not show up for walls with more friction.

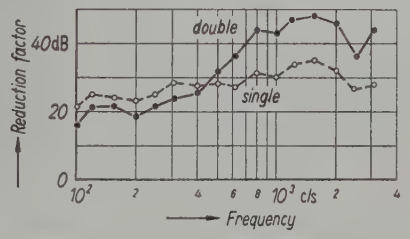


Fig. 2. Reduction factor of single and double panels of 1.3 cm plaster board. The two panels are mounted in separated brick walls; airspace = 2.5 cm, damped by rockwool blanket. Panel dimensions: 2 × 2 m².

3. Double walls with common ties (sound bridges)

In building practice it is seldom possible to employ a light double wall without connecting the panels at several points across the surface beside the edges. So far no general attempts have been made to calculate the effect of connections across the surface. Here some results of measurements carried out during the development of light panels and doors shall be given.

The measurements have been carried out on 2 × 2 m² double panels clamped to a common wall or fastened in two separate brick walls. The sound bridge consisted of a vertical wood stud with cross dimensions 5 × d cm² (d = airspace) in the midst of wall. A 2½ cm rock wool blanket was fastened to one of the panels. The mass of the walls is given in kg/m² including both panels but not the bridges.

When the panels are mounted in a common wall they are fastened to wood studs along the edges with the same cross dimensions as the vertical sound bridge.

As is well known high insulation is obtained in creating greatest possible impedance difference between neighbouring media or bodies. MEYER [5] has demonstrated that this main principle holds for sound bridges as well and has shown that in the case of lightweight panels the connecting bridges should have the highest possible impedance compared to the panel impedance i.e. they should be heavy. Iron bridges were shown to reduce the panel insulation less than wood connections. However the bridges may also have the effect of increasing the wall impedance. This change is greatest at the low frequencies where the mass reactance is low and where the wall surface influenced by the bridge is large. With very light panels, the loss in reduction factor due to the connections between the leaves may actually be compensated by the gain due to improved impedance at the wall surface.

Fig. 3 shows a summary of a number of measurements on light double walls with and without connections. As is expected the heavy panel — 70 kg/m² — is much influenced by connections at the edges. The 35 kg/m² panel is influenced in a similar way by the edge connections but further ties along the midst of the wall have little effect on the insulation. Most interesting, however, is that the 7 kg/m² panel has a higher reduction factor when a vertical bridge is introduced compared to connections only along the edges. Let us consider the insulation curve for such a case.

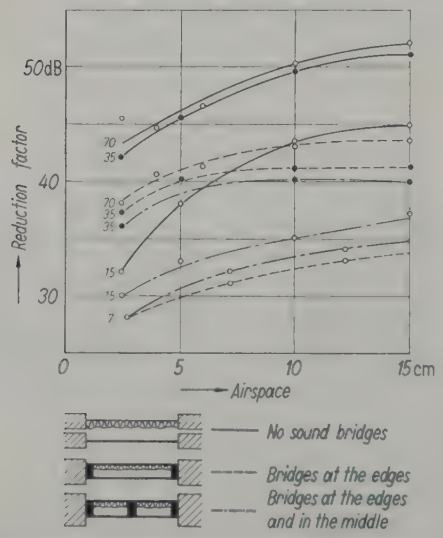


Fig. 3. Reduction factors averaged within 100 ··· 3200 c/s for some double walls with and without connections between panels. Rockwool blanket in airspace. Panel dimensions: 2 × 2 m². Cross section of sound bridge: 5 × d cm².

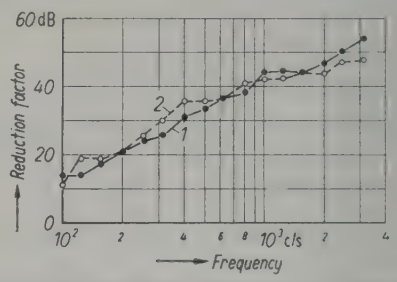


Fig. 4. Reduction factor for double 7 kg/m² wood panel mounted in common brick wall with (2) and without (1) a vertical wood sound bridge with dimensions 5 × 12.2 cm². Airspace 12.2 cm damped with rockwool blanket. Wall dimensions: 2 × 2 m².

In Fig. 4 the insulation curve is shown for the 7 kg/m² panels for the airspace 12.2 cm. The bridge increases the insulation below 800 c/s. Similar results are shown for a 21 kg/m² panel in Fig. 5. Measurements on single panels with and without supporting studs show a similar improvement of the low-frequency insulation, an effect which may be explained by the increased low-frequency wall impedance. For heavier panels this effect becomes negligible; the transmission between the panels is

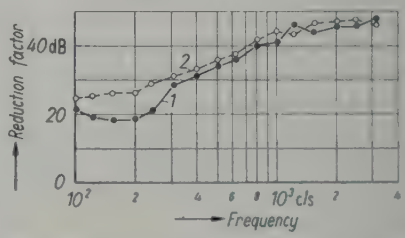


Fig. 5. Reduction factor for double 21 kg/m² wood panels mounted in separated brick walls with (2) and without (1) a wood stud (5 × 22 cm²) through the middle. Airspace: 22 cm damped with rock wool blanket. Wall dimensions: 2 × 2 m².

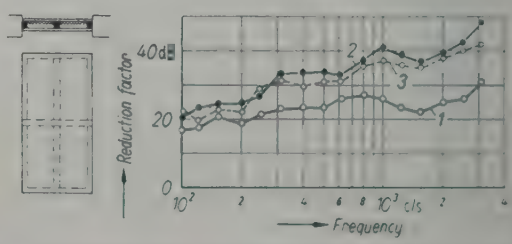


Fig. 6. Reduction factor for panels in dimensions 1 × 2 m²: 1) single panel (10.5 kg/m²); 2) double panel (21 kg/m²) without sound bridge but mounted in common brick wall; rockwool blanket in 9.3 cm airspace; 3) ditto but one vertical and one horizontal connection through the wall as sketched above. Sectional area of bridge: 9.3 × 5 cm².

enhanced because of the better impedance match between panel and stud.

For panels of smaller dimensions the edge connections are of dominating importance for the transmission between the panels. Fig. 6 shows how little a crosslike $5 \times 9.3\,\mathrm{cm}^2$ wooden bridge along the vertical and horizontal midst influences the insulation of a $1 \times 2\ \mathrm{m}^2$ panel with edge connections, a result which bears on the design of doors.

By proper mismatching of the impedances it should be possible to design sound bridges in lightweight walls in such a way that the low-frequency insulation is improved and negligible loss is met at higher frequencies.

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BERECHNUNG DER WIRKUNG VON SCHALLBRÜCKEN

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Zusammenfassung. Der Energieübertritt von Stab zu Platte für longitudinale Wellen und für Biegewellen; der Übertritt von Platte zu Platte über eine kurze stabförmige Brücke; die Verminderung der Trittschalldämmung eines schwimmenden Estrichs aus Zement und eines aus Asphalt mit hoher innerer Dämpfung durch punktweise Brücken; die Verminderung der Luftschalldämmung einer leichten Doppelwand mit punktweisen Brücken.

Summary. Energy transfer from beam to slab for longitudinal and flexural waves; transfer from slab to slab, through a short rod forming a bridge; the reduction in impact sound insulation of a floating floor made of cement and one of asphalt with high internal damping by sound bridges; reduction in airborne sound insulation of a lightweight double wall with sound bridges.

Sommaire. On étudie les questions suivantes: transmission de l'énergie d'un barreau à une plaque, dans le cas d'ondes longitudinales et d'ondes de flexion; transmission d'une plaque à une plaque par un élément court en forme de barreau et constituant un pont acoustique; diminution, du fait de ponts acoustiques locaux, de l'affaiblissement du bruit de choc sur un carrelage flottant en ciment ou en asphalte ayant un grand amortissement interne; diminution, du fait de ponts acoustiques locaux, de l'affaiblissement du bruit aérien dans une paroi double légère.

Die an Doppelwänden gemessenen Schalldämmzahlen zeigen, daß die Übertragung von Schale zu Schale hauptsächlich über die diese verbindenden Brücken erfolgt. Ebenso ist bekannt, daß selbst einzelne Brücken die Wirkung eines schwimmenden Zementestrichs sehr verschlechtern.

Es sei daher versucht, die Energieübertragung über eine stabförmige Brücke (Dichte ϱ_1, E-Modul E_1 und Länge h_1) zu berechnen, die eine Platte (ϱ_0, E_0, Dicke h_0) mit einer anderen (ϱ_2, E_2, h_2) verbindet.

Wenn in der ersten Platte eine Biegewelle den Stab trifft, so kann sie in ihm einmal eine longitudinale Welle erzeugen, andererseits auch eine Biegewelle; beide können in der zweiten Platte eine Biegewelle auslösen.

Bei der longitudinalen Welle ist das Verhältnis der von einem (unendlichen) Stab in eine (unendliche) Platte übertretenden Energie zur auffallenden einfach gekennzeichnet durch das bekannte Anpassungsgesetz:

$$\alpha_L = 1 - \left| \frac{Z_1 - Z_2}{Z_1 + Z_2} \right|^2. \qquad (1)$$

Hierin ist

$$Z_1 = A_1 \sqrt{\varrho_1 E_1}. \qquad (2a)$$

der Wellenwiderstand des Stabes (A_1 seine Querschnittsfläche) und

$$Z_2 = 2{,}3 \sqrt{\varrho_2 E_2}\, h_2^2 \qquad (2b)$$

die Impedanz der zweiten Platte bei punktweiser Anregung [1]. Besteht beides aus gleichem Material, so erfolgt ein totaler Übergang bei

$$A_1 = 2{,}3\, h_2^2, \qquad (3)$$

d.h. bei quadratischem Stab, wenn die Kantenlänge $a = 1{,}5\ h_2$, bei kreisrundem Stab, wenn der Durchmesser $d = 1{,}7\ h_2$ ist. Solche Abmessungen liegen aber gerade im Bereich des konstruktiv Üblichen. Der longitudinale Übergang ist daher im allgemeinen sehr beachtlich. Man muß schon zu sehr extremen Querschnittsunterschieden, wie zu dünnen Eisendrähten zwischen schweren Ziegelmauern greifen, um wesentliche Dämmwirkungen zu erzielen [2]. Bei Verbindung leichter Schalen kann eine Drahtverbindung der Anpassung näher kommen, als eine mit dicken und schweren Stegen. In der Tat hatte E. Meyer [3] bereits 1937 experimentell gefunden, daß Eisenstäbe zwischen Holzplatten besser dämmen als Holzstäbe, und dies durch ihre schlechtere Anpassung erklärt.

Schwieriger ist der Übertritt von Biegewellen im Stab in die Platte zu behandeln. Da hierbei die transversale Schnelle an der Übergangsstelle Null ist, wird der Energieübertritt nur noch durch das Produkt aus Moment M_2 und Winkelgeschwindigkeit w_2 bestimmt. Daher gehen auch in die zu (1) analoge Formel für den Quotienten aus übertretender zu auffalender Energie,

$$\alpha_B = 1 - \left| \frac{(1-\mathrm{i})\, Y_1 - 2\, Y_2}{(1+\mathrm{i})\, Y_1 + 2\, Y_2} \right|^2, \tag{4}$$

die Quotienten aus Winkelgeschwindigkeit und Moment ein. Und zwar bedeutet wieder Y_1 diesen Quotienten für die ankommende Welle

$$Y_1 = \frac{w_a}{M_a} = \sqrt[4]{\frac{\omega^2}{(E_1 I_1)^3 \varrho_1 A_1}}, \tag{5}$$

(ω = Kreisfrequenz und I_1 = Trägheitsmoment) und Y_2 den entsprechenden Quotienten am Stabende.

Um ihn zu berechnen, idealisieren wir die Übertragung des Momentes M_2 durch ein Kräftepaar mit dem Hebelarm e. Man erhält so ein Biegewellendipolfeld, welches sich aus zwei der früher behandelten Felder [1] mit punktförmiger Erregung formal zusammensetzen läßt. Dabei ergibt sich

$$Y_2 = \frac{3\,\omega}{E_2 h_2^3} \left\{ \frac{1}{4} + \frac{\mathrm{i}}{\pi} \left[1 - \ln \left(\frac{12\, e}{\lambda_{B2}} \right) \right] \right\} \tag{6}$$

mit der Biegewellenlänge in der Platte:

$$\lambda_{B2} = 2\,\pi \sqrt[4]{\frac{E_2 h_2^2}{12\, \varrho_2 \omega^2}}. \tag{7}$$

Im Imaginärteil geht der willkürliche Hebelarm e ein; d.h., es geht allgemein die seitliche Ausdehnung des Stabes und die Art der Verbindung ein. Trotzdem können wir Formel (4) für eine Abschätzung des Absorptionsgrades α_B benutzen, indem wir jedenfalls einen zu großen Wert erhalten, wenn wir $\mathrm{Im}(Y_2) = 0$ setzen. Unter Annahme eines quadratischen Stabes von der Kantenlänge a_1 ergibt sich dabei:

$$\alpha_{B2} < 1{,}6\, \frac{E_1}{E_2} \frac{a_1^3}{h_2^3} \frac{a_1}{\lambda_{B1}}. \tag{8}$$

Der Schallübertritt durch Momentenübertragung über die Brücke erscheint hiernach jedenfalls bei den tiefen Frequenzen gegenüber demjenigen durch Längskraftübertragung als unerheblich. Wir werden diese daher für den vorliegenden Versuch einer ersten Theorie der Schallbrücken außer Acht lassen.

Wenn die in der ersten Platte gegebene Biegewelle mit der Schnelle v_0 die Brücke trifft, entsteht an dieser eine Schnelle v_1, die sich von v_0 um den der dort rückwirkenden Kraft F_1 entsprechenden Anteil unterscheidet:

$$v_1 = v_0 - F_1/Z_0. \tag{9}$$

Die Eingangsgrößen F_1, v_1 der am Ende mit Z_2 belasteten Brücke hängen mit der gesuchten Ausgangsgröße v_2 über die bekannten Gleichungen

$$F_1 = (Z_2 \cos k_1 h_1 + \mathrm{i}\, Z_1 \sin k_1 h_1)\, v_2, \tag{10}$$

$$v_1 = \left(\mathrm{i}\, \frac{Z_2}{Z_1} \sin k_1 h_1 + \cos k_1 h_1 \right) v_2 \tag{11}$$

zusammen, worin k_1 die Longitudinalwellenzahl im Stab

$$k_1 = \frac{2\,\pi}{\lambda_{L1}} = \omega \sqrt{\frac{\varrho_1}{E_1}} \tag{12}$$

bedeutet. Im allgemeinen sind die Brücken so kurz im Verhältnis zur Wellenlänge, daß die hier enthaltenen Nullstellen des sin bzw. cos, d.h. aber Eigenschwingungen der Brücke, nicht interessieren, sondern die Brücke in (10) nur als Masse, in (11) nur als Federung wirkt:

$$F_1 = (Z_2 + \mathrm{i}\, Z_1 k_1 h_1) v_2 = (Z_2 + \mathrm{i}\, \omega \varrho_1 A_1 h_1) v_2, \tag{13}$$

$$v_1 = (\mathrm{i}\, Z_2 k_1 h_1 / Z_1 + 1) v_2 = (\mathrm{i}\, \omega h_1 Z_2 / E_1 A_1 + 1) v_2. \tag{14}$$

Die Kombination zwischen (9), (13) und (14) liefert für das interessierende Verhältnis:

$$N = \frac{v_0}{v_2} = \left(1 + \frac{Z_2}{Z_0} \right) + \mathrm{i} \left(\frac{A_1 \varrho_1}{Z_0} + \frac{Z_2}{E_1 A_1} \right) \omega h_1. \tag{15}$$

Unser technisches Ziel ist es, v_2 möglichst klein, also den Ausdruck N möglichst groß zu machen. Dabei gehen die Daten der Brücke nur in den zweiten Summanden ein. Der möglichst zu vermeidende Querschnitt ist durch das Minimum des Klammerausdrucks gegeben, also durch

$$A_1 = \sqrt{Z_0 Z_2 / E_1 \varrho_1}. \tag{16}$$

(Bei Materialgleichheit und $h_0 = h_2$ kommt das auf die Formel (3) hinaus.) Ferner ist ϱ_1 möglichst groß, E_1 möglichst klein zu wählen, doch hat das erste nur Zweck, wenn der erste Summand, also die Trägheitswirkung der Brücke, überwiegt, das zweite nur dann, wenn die Federungswirkung überwiegt. In beiden Fällen ist große Brückenlänge günstig, woraus, da jede Länge im Verhältnis zur Wellenlänge zu bewerten ist, auch folgt, daß die Dämmwirkung mit der Frequenz wächst.

Trotzdem ist der Nebenweg über die Brücken gerade bei den hohen Frequenzen von Bedeutung, weil die Dämmung über die Luftschicht noch stärker mit der Frequenz wächst.

Bei diesem Vergleich ergeben sich allerdings Unterschiede zwischen der punktweisen Trittschallerregung und der flächenhaften Luftschallerregung.

Bei der punktweisen Erregung sind das über die elastische Schicht angeregte Wellenfeld und das stets punktweise über die Brücke angeregte unmittelbar vergleichbar. Eine auf die erste Platte (Estrich) wirkende Kraft F_0 erzeugt gegenüber der Anregungsstelle in der zweiten Platte (Rohdecke) die Schnelle

$$v_2{}^* = \frac{F_0}{Z_2} \left(\frac{\omega_1}{\omega} \right)^2 = v_2{}^{**} \left(\frac{\omega_1}{\omega} \right)^2, \tag{17}$$

wobei v_2** die bei unmittelbarer Anregung der zweiten Platte sich ergebende Schnelle darstellt [4], und ω_1 die Eigenkreisfrequenz des aus der Masse der ersten Platte je Fläche und der Steife der elastischen Schicht je Fläche s bestehenden Systems

$$\omega_1 = \sqrt{s/\varrho_0 h_0}. \tag{18}$$

Befindet sich nun in der Entfernung r_0 von der Erregungsstelle eine Brücke, so hat die in (9) eingeführte primäre Schnelle dort den Wert

$$v_0 = \frac{F_0}{Z_0} \Pi (k_0 r_0), \tag{19}$$

wobei die für das rotationssymmetrische Biegewellenfeld kennzeichnende Funktion $\Pi (k_0 r_0)$ für $r_0 = 0$ den Wert 1 annimmt und für $(k_0 r_0) > 3$ in

$$\Pi (k_0 r_0) \approx \sqrt{\frac{2}{\pi k_0 r_0}}\, e^{-i\left(k_0 r_0 - \frac{\pi}{4}\right)} \tag{20}$$

übergeht. Aus v_0 ergibt sich v_2, das genau wie v_2* und v_2** ein weiteres radialsymmetrisches Biegewellenfeld in der zweiten Platte zur Folge hat.

Die durch die Brücke verminderte Verbesserung ist gegeben durch:

$$\Delta L = 10 \log \frac{(v_2{}^{**})^2}{(v_2{}^{*})^2 + v_2{}^2}$$

$$= 10 \log \left\{ 1 \Big/ \left[\left(\frac{\omega_1}{\omega} \right)^4 + \left(\frac{Z_2 \Pi}{Z_0 N} \right)^2 \right] \right\}. \tag{21}$$

In Abb. 1 ist der Frequenzgang dieses Ausdrucks aufgetragen für einen 3 cm dicken Zementestrich, der in 1 cm Abstand über einer 12 cm dicken Betondecke verlegt ist und der in 2 m Abstand von der Klopfstelle eine Brücke von 4, 1 bzw. 0,25 cm² Querschnitt aufweist. Man sieht wie selbst der letzte zu einer erheblichen „Verschlechterung" gegenüber

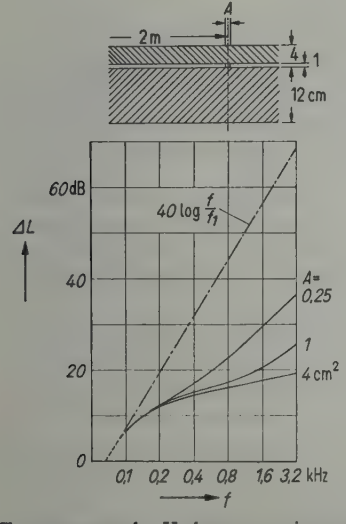


Abb. 1. Frequenzgang der Verbesserung eines schwimmend verlegten Zementestrichs ohne Brücke (strichpunktiert), bzw. mit A cm² großer Brücke in 2 m Abstand von der Klopfstelle (ausgezogen).

der ohne Brücke sich ergebenden geraden Linie führt.

Wesentlich günstigere Ergebnisse ergeben sich bei Asphaltestrichen mit unfreiwilligen Brücken, vor allem deshalb, weil sie eine große innere Dämpfung mit einem Verlustfaktor, d.h. Verhältnis der verlorenen zur elastischen Formänderungsarbeit von etwa $\eta = 0{,}2$, aufweisen. Dadurch sinkt die Amplitude mit der Entfernung vom Erregungsort mit:

$$|\Pi| \approx \sqrt{\frac{2}{\pi k_0 r_0}}\, e^{-\eta k_0 r_0/4}. \tag{22}$$

In Abb. 2 ist unter Berücksichtigung dieser Dämpfung der Frequenzgang von ΔL aufgetragen für einen 2,5 cm starken Asphaltestrich, der wieder in 1 cm Abstand über einer 12 cm dicken Betondecke verlegt ist und der eine 1 cm² große Asphaltbrücke in 0, 1, 2 bzw. 4 m Abstand aufweist. Im letzten Fall erhält man überhaupt keine Abweichung gegenüber der idealen Verbesserung.

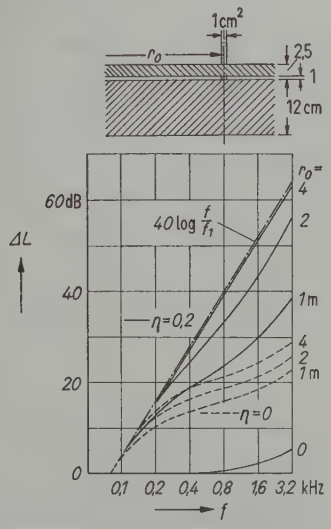


Abb. 2. Frequenzgang der Verbesserung eines schwimmend verlegten Asphaltestrichs ohne Brücke (strichpunktiert) bzw. mit 1 cm² großer Brücke in verschiedenen Abständen ohne (gestrichelt) und mit (ausgezogen) Berücksichtigung des Verlustfaktors.

Bei der Luftschalldämmung spielt der Ort der Brücke keine Rolle, da hier die ganze Wand durch den Luftschalldruck p_0 vor der Wand zu erzwungenen Biegewellen erregt wird. Würde der Schall unmittelbar unter dem Einfallswinkel ϑ_0 auf die zweite Platte auftreffen, so würde sich unter der Annahme, daß auch sie eine hohe Grenzfrequenz aufweist [5], ergeben:

$$v_0{}^{**} = \frac{p_0 \cos \vartheta_0}{i \omega m_2}. \tag{23}$$

Nach Vorsetzen der ersten Platte müßte sich diese Amplitude ohne Brücken verringern lassen auf [6]

$$v_2^* = v_2^{**} \left(\frac{\omega_1}{\omega}\right)^4; \tag{24}$$

und für die Schnelleamplitude in der ersten Platte gilt dabei:

$$v_0 = \frac{p_0 \cos \vartheta_0}{i\,\omega m_0}. \tag{25}$$

Damit läßt sich auch nach (15) die Schnelle v_2 im Fußpunkt einer Brücke errechnen.

Nun ist aber zu bedenken, daß diese Schnelle zu einem ganz anderen Wellenfeld gehört, das auch zu einer ganz anderen Abstrahlung führt. Indem wir diesen Unterschied durch einen Strahlungsfaktor Γ berücksichtigen, nimmt der Ausdruck für die durch die Brücken verminderte Verbesserung der Schalldämmzahl R hier die Form an:

$$\Delta R = 10 \log \frac{(v_2^{**})^2}{(v_2^*)^2 + (\Gamma v_2)^2}$$

$$= 10 \log \left\{1 \Big/ \left[\left(\frac{\omega_1}{\omega}\right)^4 + \left(\frac{m_2}{m_0}\frac{\Gamma}{N}\right)^2\right]\right\}. \tag{26}$$

Der Strahlungsfaktor ist jedenfalls eine Funktion der Frequenz, allgemeiner ihres Verhältnisses zu der von dem hiermit verwandten Koinzidenzeffekt her bekannten Grenzfrequenz [5], bei welcher die Länge der freien Biegewelle gleich der Luftwellenlänge ist:

$$\omega_g = (c_a^2/h_0)\sqrt{12\,\varrho_0/E} \tag{27}$$

(c_a = Schallgeschwindigkeit in Luft).

Obschon die quantitative Erfassung des Strahlungsfaktors noch im Fluß ist, kann man auf Grund der vorliegenden experimentellen Erfahrungen [7], [8] und theoretischen Abschätzungen [9] ··· [11] schon heute sagen, daß er für $\omega > \omega_g$ nicht sehr verschieden von 1 ist, aber für $\omega < \omega_g$ sehr viel kleiner werden kann. Es ist also zur Herabsetzung der Wirkung von Schallbrücken günstig, wenn ω_g möglichst hoch liegt, d.h. wenn die zweite Platte schwer, aber wenig steif ist. Bei tragenden Decken ist das kaum zu erreichen, wohl aber bei dünnen Leichtbauplatten und zwar durch dieselben Maßnahmen, die zur Vermeidung des Koinzidenzeffektes dienen, also beispielsweise durch Rillung der Platten [5].

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[3] Meyer, E., Akust. Z. 2 [1937], 74.
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DER EINFLUSS DER BIEGESTEIFIGKEIT AUF DIE SCHALLDÄMMUNG VON DOPPELWÄNDEN *)

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Zusammenfassung. Doppelwände aus dünnen Wandschalen weisen häufig eine größere Schalldämmung auf als Doppelwände aus dicken, steifen Schalen, obwohl diese Wände um ein Mehrfaches schwerer sind als die erstgenannten. Das günstige Verhalten der Wände mit dünnen Schalen ist auf die geringe Schallabstrahlung des übertragenen Körperschalls unterhalb der Grenzfrequenz der Wandschale zurückzuführen. Zwei Beispiele für zweckmäßig ausgeführte schalldämmende Leichtwände werden mitgeteilt.

Summary. Double walls of thin leaves frequently give better sound insulation than those with thick stiff leaves, although the latter are much heavier. The most advantageous property of walls with thin leaves is to be ascribed to the small radiation of structureborne sound below the critical frequency of the leaf. Two practical examples are given of designed sound-insulating lightweight partitions.

Sommaire. Il arrive souvent que des parois doubles en feuilles minces produisent un plus grand isolement sonore que des parois doubles en feuilles épaisses et rigides, bien que celles-ci soient bien plus lourdes que les premières. Cette qualité des parois à feuilles minces est due au faible rayonnement acoustique au-dessous de la fréquence limite de la feuille. On donne deux exemples de réalisation appropriée de parois légères isolantes.

1. Vergleich von Wänden mit verschieden steifen Schalen

Untersucht man im Laboratorium die Luftschalldämmung von Doppelwänden aus verschieden dicken und damit verschieden steifen Wandschalen, dann stellt man fest, daß die Dämmung durchaus nicht mit der zunehmenden Masse der Wände (je Flächeneinheit) größer wird, sondern häufig ab-

*) Diese Untersuchungen sind im Auftrag der Forschungsgemeinschaft Bauen und Wohnen in Stuttgart durchgeführt worden.

nimmt. Es wurde einmal eine Doppelwand aus 1,2 cm Gipsplatten (sogen. Rigipsplatten), zum anderen aus 8 cm Gipsplatten (mit eingelegten Holzwolleplatten) untersucht. Die erstgenannte Doppelwand hatte ein Flächengewicht von 22 kg/m², die zweite ein solches von 140 kg/m². Beide Wände waren in die 2,5 m × 3 m große Öffnung einer Wand aus 25 cm Vollziegeln eingebaut, die die eine Schale einer Doppelwand zwischen zwei Hallräumen bildete. Die Schalldämmung der schweren Wand war, wie die Abb. 1b und 2b zeigen, zwischen 300 und 2000 Hz kleiner als die der leichten Doppelwand. Beide Doppelwände sind zwar deutlich besser als die entsprechenden einschaligen Wände der halben Masse (Kurven a).

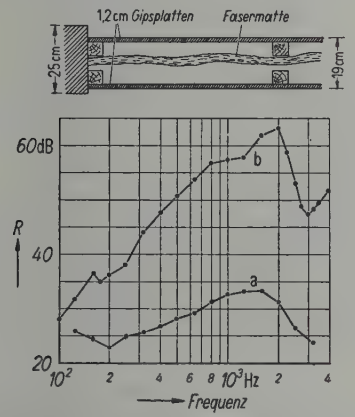


Abb. 1. Schalldämmung einer Doppelwand aus 1,2 cm Gipsplatten.
a) eine Schale (10 kg/m²); b) Doppelwand (22 kg/m²).

Die naheliegende Frage ist zunächst, ob die relativ große Schallübertragung bei der schweren Wand über die Luftschicht oder als Körperschall über die gemeinsame Einspannung erfolgt. Dazu wurde diese Wand so aufgebaut, daß die beiden Wandschalen in körperschallmäßig getrennte Maueröffnungen eingebaut wurden (siehe Abb. 2c). Bei sonst gleichem Aufbau ergibt sich nun eine erhöhte Dämmung (siehe Kurve c in Abb. 2).

Die Schallübertragung erfolgt somit über die gemeinsame Einspannung. Die Schwingungsamplituden der Wandschalen auf der leisen Seite der beiden Doppelwände (Abb. 1b und 2b) ergaben sich bei der Messung als ungefähr gleich groß. Der Unterschied in der Dämmung kann deshalb nur auf das verschiedenartige Abstrahlverhalten der Schalen zurückzuführen sein.

Für den Nachweis soll ein Abstrahlfaktor s eingeführt werden [1]) (siehe auch [5]).

[1]) Dieses so definierte s entspricht bis auf einen Faktor 2 dem „coefficient de rayonnement" von BRILLOUIN [3].

$$s = \frac{N}{\frac{1}{2} c \varrho v^2 S},$$

wobei bedeuten:

N = in den „leisen" Raum abgestrahlte Schall-Leistung der Doppelwand,
ϱ = Luftdichte,
c = Schallgeschwindigkeit in Luft,
v = Schnelleamplitude der Biegeschwingungen der Wandschale,
S = Fläche der Wandschale.

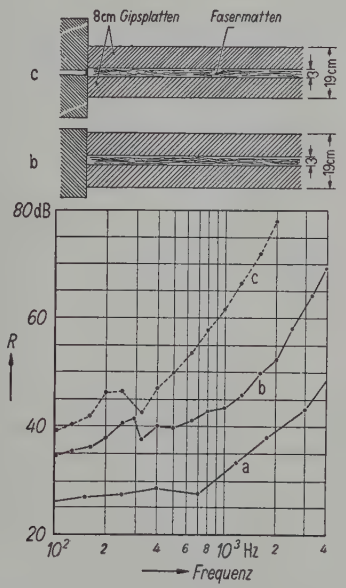


Abb. 2. Schalldämmung einer Doppelwand aus 8 cm Gipsplatten.
a) eine Schale (70 kg/m²); b) Doppelwand, gemeinsame Einspannung der Wandschalen; c) Doppelwand, Wandschalen in getrennten Rahmen (Körperschallübertragung unterbunden).

Dabei ist die tatsächlich abgestrahlte Leistung N in Beziehung gesetzt zu der Abstrahlung eines gleich großen Flächenstücks innerhalb einer unendlich großen Fläche, deren Flächenelemente konphas schwingen und zwar mit derselben Amplitude und Frequenz wie die betrachtete Doppelwandschale.

Abb. 3 zeigt den Abstrahlfaktor s, in logarithmischem Maß, für die beiden untersuchten Doppelwände. Für die 8 cm Gipsplatten ergibt sich oberhalb 300 Hz ein Abstrahlfaktor von 1 bis 1,6, während die 1,2 cm Gipsplatten unterhalb 2000 Hz einen Wert von rund 1/100 bis 1/10 aufweisen. Entsprechend den Ergebnissen von CREMER [2], BRILLOUIN [3] und GÖSELE [4] zeigen die Wandschalen unterhalb ihrer Grenzfrequenz f_{gr} eine anomal niedrige Schallabstrahlung, während darüber die Abstrahlung nahezu der einer konphas schwingenden Platte entspricht. Bei den Werten der Abb. 3 ist zu beachten, daß die Schwingungen der Wand-

schalen sich teils aus freien (durch Körperschall-Übertragung), teils aus erzwungenen (durch Luftschall-Übertragung) Biegeschwingungen zusammensetzen. Die gemessenen Werte stellen somit nicht das Abstrahlverhalten für freie Biegewellen dar. Infolge der höher liegenden Grenzfrequenz der 1,2 cm dicken Gipsplatten ist deren Abstrahlung im hauptsächlichen Frequenzbereich wesentlich geringer als die der steiferen 8 cm Gipsplatten. Darauf ist in erster Linie die bessere Dämmwirkung der leichten gegenüber der schweren Doppelwand zurückzuführen.

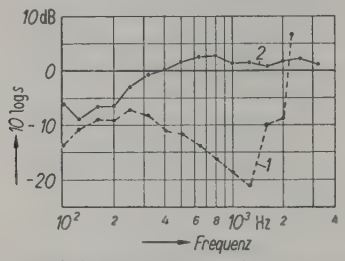


Abb. 3. Abstrahlfaktor der Wandschalen von Abb. 1b (Kurve 1) und Abb. 2b (Kurve 2).

2. Praktische Folgerungen

Will man die Schalldämmung von leichten Doppelwänden, die eine gemeinsame Einspannung haben, vergrößern, so sollte die Masse (je Flächeneinheit) der Wandschalen erhöht werden. Damit ist jedoch im allgemeinen eine erhöhte Biegesteifigkeit und eine Erniedrigung der Grenzfrequenz verbunden. Die Abstrahlung des übertragenen Körperschalls wird dadurch erhöht, wodurch die gewünschte Wirkung in ihr Gegenteil verkehrt wird. Anzustreben ist eine Erniedrigung der Steifigkeit (z.B. Einsägen von Rillen [1]).

Eine andere Möglichkeit besteht darin, die beiden Wandschalen verschieden auszuführen, und zwar die eine relativ schwer, die andere dagegen dünn und mit entsprechend hoher Grenzfrequenz. Die Abb. 4 und 5 zeigen zwei Beispiele, wie eine dünne, biegeweiche Wandschale an einer steiferen Wandschale befestigt werden kann. Im einen Fall (Abb. 4) ist eine zweite Schale aus 5 cm Holzwolle-Leichtbauplatten vor der zu verbessernden Wand aufgebaut, wobei die Holzwolleplatten mit einzelnen Nägeln über eine weiche Zwischenschicht an der Wand befestigt sind. Im zweiten Fall (Abb. 5, Kurve

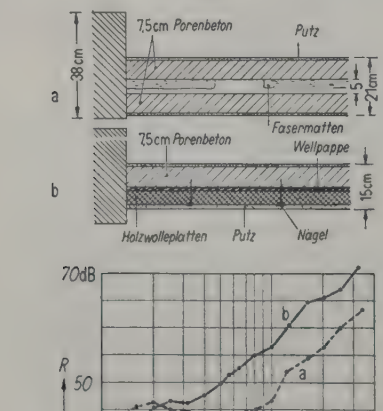


Abb. 4. Verbesserung einer Leichtwand aus Porenbeton durch eine zweite vorgesetzte Schale.
a) zweite Schale aus 7,5 cm Porenbeton; b) zweite Schale aus Holzwolle-Leichtbauplatten.

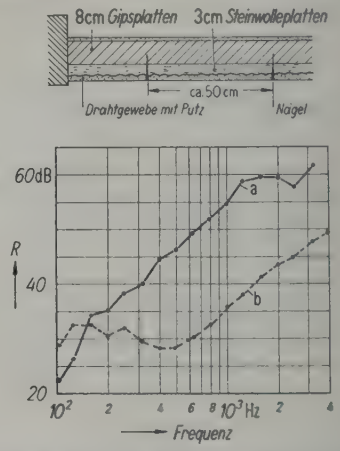


Abb. 5. Verbesserung einer Leichtwand durch eine Putzschale auf 3 cm Steinwolleplatten.
a) mit Putzschale; b) Einfachwand ohne Putzschale.

a) ist ein Drahtgewebe als Putzträger über 3 cm dicke Steinwolleplatten an der zu verbessernden Wand (Kurve b) mit einzelnen Nägeln befestigt und verputzt. Der Abstrahlfaktor s für diese Wandschalen ist in Abb. 6 angegeben. Trotz einer zusätzlichen Körperschall-Übertragung über einzelne Nägel zwischen den beiden Schalen ist die Verbesserung der Schalldämmung in beiden Fällen gut.

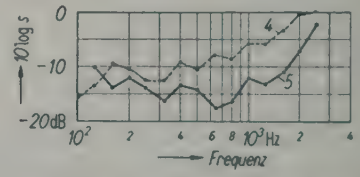


Abb. 6. Abstrahlfaktor der dünnen Wandschalen von Abb. 4b (Kurve 4) und Abb. 5a (Kurve 5).

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[5] Gösele, K., Acustica 3 [1953], 243.

ÉTUDE THÉORIQUE ET EXPÉRIMENTALE DES PAROIS MINCES ET APPLICATION AU TRAITEMENT ACOUSTIQUE DES FUSELAGES D'AVION

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Sommaire. Étude de structures légères utilisées en construction aéronautique, constituées de matériaux poreux, associés à des membranes plastiques, couplées au revêtement métallique. On montre que l'affaiblissement aux fréquences aiguës est très supérieur à celui donné par la loi de masse. On décrit l'installation de mesure donnant directement la courbe de transparence acoustique, et on expose une méthode approchée de calculs de la réduction additive des niveaux sonores dans une cabine d'avion.

Zusammenfassung. Es werden aus porösen Materialien bestehende leichte Strukturen für Flugzeugkonstruktionen untersucht, die auf Kunststoffplatten angebracht und hiermit an metallischen Verkleidungen befestigt sind. Es wird gezeigt, daß die Dämmung bei hohen Frequenzen erheblich höher ist als es nach dem „Massengesetz" zu erwarten wäre. Es wird die Meßanordnung beschrieben, die direkt die akustische Durchlaßkurve ergibt und es wird eine Näherungsmethode angegeben, die dadurch verursachte zusätzliche Herabsetzung des Schalldruckpegels in einer Flugzeugkabine zu berechnen.

Summary. Light structures for aircraft construction are studied. They consist of porous materials associated with plastic sheets joined to the metallic skin. The insulation at high frequencies is shown to be very superior to that given by the mass law. The experimental equipment is described. It gives directly the curve of acoustic transmission. An approximate method is given for the calculation of the additional decrease of sound level in an aircraft cabin.

Les structures utilisées pour insonoriser les cabines d'avions sont constituées par des "couplages" composées de matériaux poreux tels que feutres de verre, Kapok ignifugé etc... associés à des membranes imperméables en amiante, résines polyvinyliques et autres.

Les limitations sévères de la surcharge de l'avion, (le poids global de l'insonorisation ne devant excéder 4 à 5 kg par passager) rendent particulièrement nécessaire la mesure en laboratoire du coefficient de transmission et du coefficient d'absorption à la réflexion des structures.

Nous présentons quelques structures types, dont le coefficient de transmission exprimé sous sa forme usuelle d'affaiblissement en dB, au dessous du niveau de l'énergie incidente a été mesuré en fonction continue de la fréquence (0,05 ··· 10 kHz) pour des couplages d'une épaisseur globale de 70 mm.

La surcharge a été rendue variable depuis 0,360 kg jusqu'à 1,460 kg par mètre carré. Les résultats des mesures ainsi que la composition des structures sont visibles sur les figures ci-dessous.

La Figure 1 indique la transmission acoustique d'une plaque plane en duralumin dont la masse par unité de surface est de 2,8 kg/m². Son affaiblissement calculé d'après la relation connue

$$I = 10 \log [1 + (\omega m/\varrho c)^2]$$

est également donné sur la même figure.

L'accord est très satisfaisant pour les fréquences supérieures à 1000 Hz. L'accord est excellent au dessus des fréquences principales de résonance de la paroi.

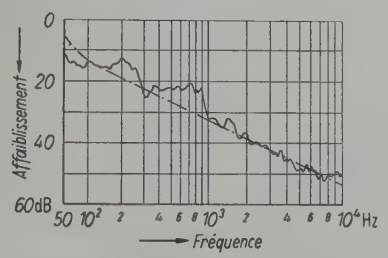


Fig. 1. Transmission acoustique d'une plaque plane de duralumin densité 2,8 kg/m².

Les figures suivantes se rapportent à la même plaque couplée avec des matériaux insonorisants.

On sait que le confort acoustique des cabines d'avions dépend du niveau de bruit de masque, des fréquences de la parole. C'est pourquoi nous classons les structures pour 3000 Hz (Figs. 2, 3 et 4):

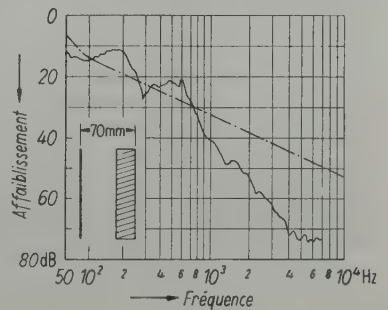


Fig. 2. Transmission acoustique d'une cloison: plaque dural plus intervalle d'air plus feutre fibre de verre superfine — surcharge 0,36 kg/m² (en plus du poids de la tôle).

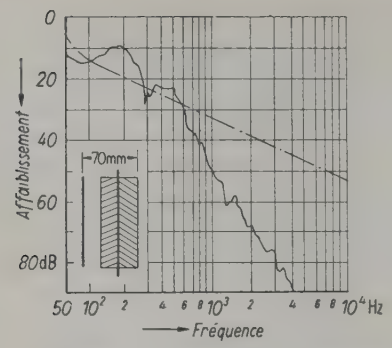


Fig. 3. Comme Fig. 2 — Addition d'une membrane polyvinylique dans l'épaisseur du feutre. Surcharge 0,91 kg/m².

Structure 1, surcharge 0,360 kg/m², gain par rapport à la loi de la masse, approximativement 20 dB; structure 2, surcharge 0,910 kg/m², gain de l'ordre de 30 dB; structure 3, surcharge 1,46 kg/m², gain de l'ordre de 35 dB.

Les parois légères deviennent particulièrement efficaces pour la réduction des fréquences moyennes et hautes de bruit, lorsque les matériaux poreux sont associés à une membrane imperméable. L'adjonction d'une seconde membrane ne conduit pas à une réduction "économique" du bruit.

Il est approximativement possible de calculer la réduction additive des niveaux sonores résultant de l'insonorisation des cabines d'avion à partir des mesures en laboratoires des deux paramètres spécifiques des structures sélectionnées.

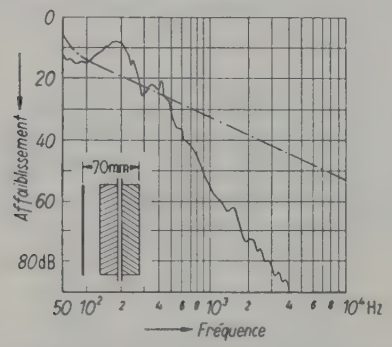


Fig. 4. Comme Fig. 2 — Addition de deux membranes espacées de 12 mm. Surcharge 1,46 kg/m².

La théorie élémentaire conduit à l'expression de l'isolement phonique d'une enceinte fermée entourée d'un champ sonore isotrope et homogène qui est:

$$I = 10 \log \frac{1}{\tau + r\tau + r^2\tau + \cdots} = 10 \log \frac{a}{\tau},$$

τ étant le coefficient de transmission, r et a respectivement les coefficients de réflexion et d'absorption de l'intérieur.

Le niveau sonore peut être réduit en appliquant des matériaux insonores à l'intérieur. En tenant compte du fait que les différentes parties de l'intérieur peuvent avoir des coefficients différents et en appelant les nouvelles valeurs des coefficients τ' et a', on calcule pour la réduction additive, résultant du revêtement,

$$I_{ad} = 10 \log \frac{\Sigma S_i a_i'}{\Sigma S_i a_i} \frac{\Sigma S_i \tau_i}{\Sigma S_i \tau_i'} \qquad (1)$$

Bien qu'à l'extérieur du fuselage d'un avion, le champ acoustique soit hétérogène le calcul de la réduction additive des niveaux sonores dans la cabine est approximativement possible par la formule:

$$I_{ad} = 10 \log \frac{\Sigma S_i a'_i}{\Sigma S_i a_i} \frac{\Sigma S_i \tau_i}{\Sigma S_i \tau_i'} \left[\frac{E'(\omega)}{E(\omega)}\right]^r \qquad (2)$$

$E'(\omega)$ étant l'énergie sonore absorbée au cours de sa propagation longitudinale dans la cabine insonorisée depuis la zone de densité spectrale énergétique maximum (cône d'échappement des réacteurs, plan des hélices), $E(\omega)$ l'énergie absorbée dans la cabine nue et r la distance en mètres de cette zone au point d'observation.

La figure 5 indique l'exemple d'application du calcul conduit d'après (2) dans le cas d'un biréacteur de transport des passagers.

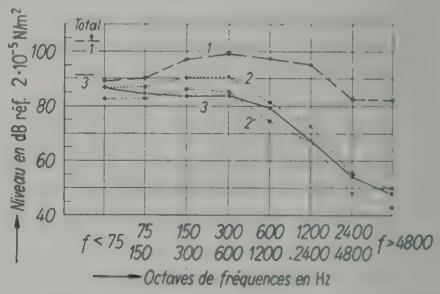


Fig. 5. Niveaux sonores dans une cabine de passagers (extrémité arrière) palier à l'altitude 3 200 m. Poussée 700 kg par réacteur.
1) Avant insonorisation.
2 et 2') Limites calculées des niveaux de bruit après traitement acoustique.
3) Niveaux sonores mesurés après traitement.

La courbe 1 donne les niveaux sonores en vol dans la cabine nue, les courbes 2 et 2' les limites calculées de la réduction additive des niveaux sonores résultant de l'insonorisation de la cabine. La courbe 3 qui indique les niveaux sonores mesurés est en bon accord avec les prédictions du calcul.

L'installation pour la mesure du coefficient de transmission des structures réalisée en notre laboratoire est constituée essentiellement par une cabine totalement absorbante dont la face avant est percée par une baie de 1 m par 1 m et de 0,25 m de profondeur. L'échantillon à l'étude est serré dans la baie grâce à la compression de ressorts à lames obtenue

par la fermeture d'un cadre rigide à deux battants (Fig. 6 et 7).

La source sonore est constituée par l'association d'une batterie de haut-parleurs et d'un ionophone. Deux microphones de mesure (électrostatiques) sont fixés sur des supports orientables de part et d'autre de la structure. Les deux tensions microphoniques subissent chacune une transformation logarithmique et sont appliquées à chaque plaque déflectrice d'un tube cathodique. Le balayage est effectué en fonction du logarithme de la fréquence d'essai.

L'appareil d'enregistrement indique donc directement la courbe de variation de la transmission acoustique en décibels donnée par:

$$T = 20 \log \frac{p_1}{p_2}.$$

La sensibilité de l'appareil permet de déceler des différences de niveaux en fonction de la fréquence

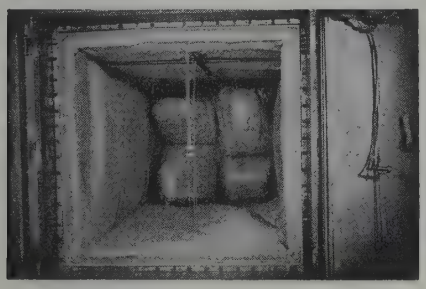


Fig. 6. Vue de la cabine absorbante montrant le microphone fixé sur le cadre orientable et le dispositif de serrage de l'échantillon par cadre à ressorts comprimés.

atteignant 100 dB. L'erreur instrumentale est inférieure à $\pm$ 2 dB. La durée d'enregistrement est une minute.



Fig. 7. Face avant de la cabine — montage d'un hublot — batterie de haut-parleurs avec ionophone au centre.

Dans un ordre de technique différent, M. BRILLOUIN, a effectué, à la demande d'un organisme officiel, des mesures de transmission à travers des cloisons démontables.

Les essais ont donné un affaiblissement moyen de l'ordre de 20 dB aux très basses fréquences, 25 dB aux moyennes et 30 dB aux aiguës.

Discussion

KIRSCHNER, F.: a) Vous avez exploré le champ acoustique par un microphone qui avait trois degrés de liberté. Les valeurs données dans les graphiques sont-elles des valeurs moyennes?

b) Ou bien faut-il en déduire que les ondes sont planes après avoir traversé les différents échantillons?

CONFÉRENCIER: a) Ces valeurs sont des valeurs exactes.

b) Oui, les ondes restent sensiblement planes.

SOME FUNDAMENTAL MEASUREMENTS ON SINGLE AND DOUBLE PLATE STRUCTURES

by V. M. A. PEUTZ

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Summary. Measurements on the transmission loss of plates of wood, lead, rubber and aluminium in a reverberant sound field are discussed. In agreement with the results of LONDON a.o. a rather high insulation is found for the low frequencies. The transmission loss of a double wall is compared with that of the corresponding single wall.

Sommaire. On donne les résultats de mesures de l'isolement acoustique de panneaux de bois, de plomb, de caoutchouc et d'aluminium; les mesures ont été faites dans un champ sonore réverbérant. On trouve en accord avec les résultats de LONDON e.a. une assez grande valeur de l'isolement acoustique aux basses fréquences. On compare aussi l'isolement d'une paroi double et celui de la paroi simple correspondante.

Zusammenfassung. Es werden Messungen der Schalldämmzahl von Platten aus Holz, Blei, Gummi und Aluminium diskutiert. In Übereinstimmung mit den Ergebnissen von LONDON u.a. fand man eine verhältnismäßig große Isolation für tiefe Frequenzen. Die Schalldämmung einer Doppelwand wird mit der einer entsprechenden einfachen Wand verglichen.

A brief account of the results of some measurements and of theoretical considerations concerning single and double plate structures is presented.

1. Single plates

There are as yet only a few published results of insulation measurements on thin plates and these are not in good agreement with theory.

Fig. 1 gives results of A. LONDON [1] and of the T.P.D. *) (unpublished). At low frequencies the transmission loss is above that predicted by mass law, and has a minimum at circa 200 c/s.

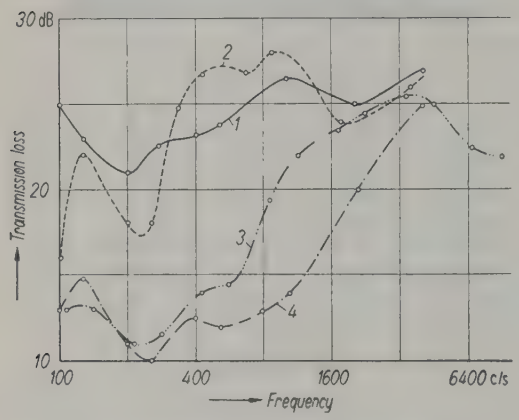


Fig. 1. The transmission loss versus frequency.
(1) 12.7 mm plywood (LONDON [1]);
(2) 12.7 mm plywood + 0.4 mm aluminium (LONDON [1];
(3) 6 mm plywood 3.6 kg/m² (T.P.D. unpublished);
(4) 0.4 mm aluminium (LONDON [1]).

At high frequencies there is neither agreement with theory, nor between the various results. Only the curve for 0.4 mm aluminium given by LONDON follows the mass law for random incidence at the higher frequencies in lying almost 6 dB below the calculated curve for normal incidence (see KOSTEN Fig. 1, p. 264), the slope being ca. 5.5 dB/octave.

On account of leakages, of clamping difficulties and of the properties of the reverberant rooms such as distribution of absorption material etc., it is difficult to get reproducible and reliable measurements and the true corrections for those measurements.

In order to correct the results of our measurements we adopted first the usual method in which the correction follows from the reverberation time of the receiving reverberant room. It was found, however, that referring to the transmission loss measured between the two reverberant rooms when there was no wall between them, the case of "zero insulation", gave better results. Only then we got reproducible values for the transmission loss of the plates independent of the distribution of the absorption as well as of the dimensions of the reverberant rooms. Measurements on plates of lead, rubber, aluminium and plywood will be described (Fig. 2 ··· 6).

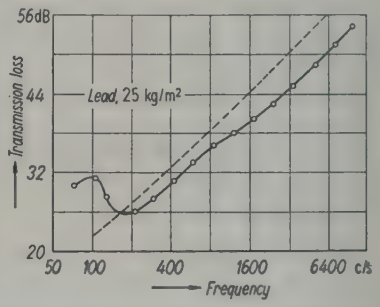


Fig. 2. The transmission loss versus frequency for lead 25 kg/m²; the dotted line gives the transmission loss as predicted by the mass law for normal incidence.

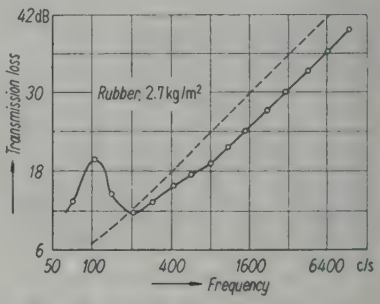


Fig. 3. As Fig. 2; rubber 2.7 kg/m².

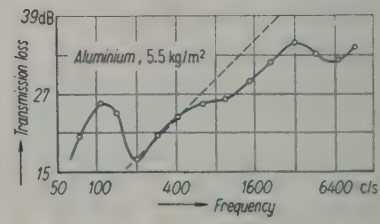


Fig. 4. As Fig. 2; aluminium 5.5 kg/m².

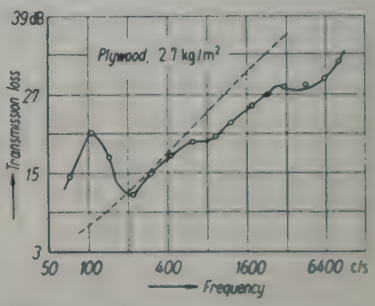


Fig. 5. As Fig. 2; plywood 2.7 kg/m².

*) Technical Physics Department T.N.O. and T.H. Delft.

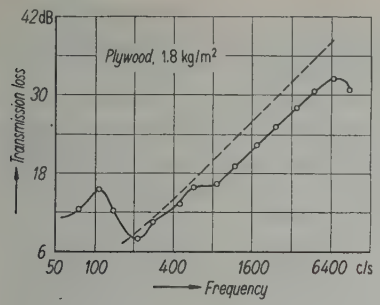


Fig. 6. As Fig. 2; plywood 1.8 kg/m².

We always find the same general features (Fig. 7). Above a certain frequency, usually ca. 800 c/s, the curve of the transmission loss versus frequency agrees with theory for random incident noise up to 80 to 83 degrees. The curve lies about 6 dB below the mass law curve for normal incidence, the slope being ca. 5.5 to 6 dB/octave.

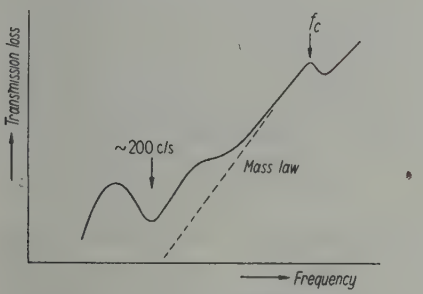


Fig. 7. General features of the transmission loss versus frequency; the dotted line gives the transmission loss as predicted by the mass law for sound incident at angles up to ca. 80°.

Below the mentioned frequency the transmission loss is greater than that expected on theoretical grounds. Down to 200 c/s this improvement depends on the material, being greatest with aluminium and smallest with rubber. The lower the frequency is the smaller the transmission loss and at 200 c/s there is a minimum in the transmission loss. From there the transmission loss rises again until a maximum is reached at 100 c/s, followed by a rapid fall. This maximum at 100 c/s and the minimum at 200 c/s were found to be *independent* of the material; they also occur at the same frequencies in the curves of London and of the T.P.D.

A heavy mass placed at the centre on the wall, which was tested in a horizontal position, did influence this maximum and minimum. We see a similar displacement of the maximum and minimum to higher frequencies in the results obtained by the T.P.D., where an extra stud was used (Fig. 8).

It is not possible to explain the insulation improvement merely with the aid of the equation of motion of the wall. Even if in addition we take into account the tension of the plate and the higher order terms, we do not find — at least not above ca. 50 c/s — at the intensities used measurable nonlinear phenomena that could explain the high transmission loss at low frequencies.

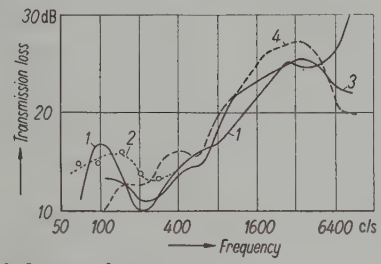


Fig. 8. Influence of extra support.
(1) 5 mm plywood;
(2) idem with mass placed at the centre;
(3) 6 mm plywood (T.P.D. unpublished);
(4) idem with support (T.P.D. unpublished).
(The differences between 3 and 4 at high frequencies are due to leakages.)

So far we did not take into account that the walls have finite dimensions. One might expect indeed an abnormal transmission loss at the low frequencies due to this fact but only, as Schoch has shown (see p. 266), in the neighbourhood of the fundamental resonance frequency. In our case the highest fundamental resonance frequency was approx. 30 c/s and generally it is even much lower, so that the finite dimensions cannot explain the higher transmission loss measured.

We cannot explain the low-frequency improvement of the transmission loss from the natural frequencies of the reverberant rooms either, since the improvement occurs independent of the room size (Fig. 1). Nor did we find any indication of standing waves in our reverberant rooms, which is to be expected of course when white noise is used, as it was in fact.

2. Double plates

The behaviour of a double wall in a random incident sound field is not fully understood either. We will give here some results of a theory that will be published shortly.

The transmission loss of a double wall can be calculated from the impedances Z_1 and Z_2 of the two constituent single walls.

Let

$$D = 1 + \frac{Z}{\varrho c} \cos\theta,$$

$$k_x = \frac{\omega}{c} \cos\theta,$$

θ = angle of incidence of the oblique plane wave,
d = thickness of air space.

Then we have for the energy transmission ratio T_θ, $k_x d$ being small,

$$T_\theta \approx \tfrac{1}{4}\,\alpha^2\,|\,D_1 + D_2\,|^2$$

where

$$\alpha = \left|\,1 + \mathrm{j}\,\frac{\omega d}{c}\,\frac{D_1 D_2}{D_1 + D_2}\cos\theta\,\right|.$$

We see that the transmission ratio of the double wall differs by the factor α^2 from that of the corresponding single wall, i.e. the double wall with an airspace $d = 0$. The resulting impedance is then $Z_1 + Z_2$, the value for D being $D_1 + D_2$. The insulation of the corresponding single wall will therefore be equal to that of a single wall with an impedance $(Z_1 + Z_2)$.

We will now examine α^2. For resonance α will have a minimum. We will confine ourselves here to two equal walls with a pure mass impedance; then we have resonance if

$$\tfrac{1}{2}\,\omega^2\,\frac{md}{\varrho c^2}\cos^2\theta = 1,$$

whence

$$\omega_{n\,res} = \omega_{\theta\,res}\cos\theta_{res},$$

where $\omega_{n\,res}$ is the angular resonance frequency for normal incidence, $\omega_{\theta\,res}$ the same for an angle of incidence θ.

For resonance the value of α is a minimum:

$$\alpha_{res} = \frac{\varrho c}{\omega_{\theta res}\, m\cos\theta} = \frac{\varrho c}{\omega_{n\,res}\, m},$$

which is thus independent of θ and ω.

Now we can write:

$$\alpha^2 = \left[1 - \left(\frac{\cos\theta}{\cos\theta_{res}}\right)^2\right]^2 + \alpha^2_{res}\left(\frac{\cos\theta}{\cos\theta_{res}}\right)^4.$$

We see that for increasing θ_{res} the resonance becomes sharper, and the overall value of α^2 beyond resonance greater.

Taking all this into account and computing the transmission loss for random incident noise we find that the infinite double wall certainly has a higher transmission loss than the corresponding single wall for frequencies above $\omega = 8\,\omega_{n\,res}$.

It would be advantageous to develop constructions in such a way that this dip falls within the frequency range where single walls have their unexplained low-frequency maximum, thus providing a better insulation characteristic.

In case the two walls are not equal we get similar formulae with almost the same conclusions.

Our measurements are in quite good agreement with theory (Fig. 9). Above resonance at normal incidence the transmission loss remains for 1 to 1.5 octave below that of the corresponding single wall, as was to be expected.

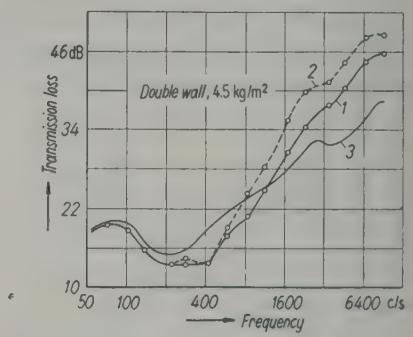


Fig. 9. Transmission loss versus frequency of double wall (4.5 kg/m²),
$d = 15$ mm, $m_1 = 1.8$ kg/m², $m_2 = 2.7$ kg/m².
(1) double wall;
(2) idem, with rockwool at the edges;
(3) computed corresponding single wall.

References

[1] London, A., J. Res. nat. Bur. Stand., Wash. 42 [1949], 605. Res. Paper 1998.
[2] London, A., J. acoust. Soc. Amer. 22 [1950], 270.

ISOLEMENT PHONIQUE DE QUELQUES STRUCTURES LÉGÈRES

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Sommaire. Les auteurs ont étudié quelques types de structures légères d'un poids maximum de 15 kg/m². Les échantillons étaient de 2 m² environ. Pour une paroi simple il a été trouvé des isolements différant sensiblement de la loi de masse. Pour des structures complexes, d'importantes améliorations ont été notées, qu'il s'agisse de structures à base de duralumin, bois compressé, ou de laine de roche. Un mode de fixation entièrement élastique a été utilisé.

Zusammenfassung. Die Autoren haben einige Typen von leichten Wänden vom Maximalgewicht 15 kg/m² untersucht. Die Proben hatten eine Größe von etwa 2 m². Für die einfache Wand fand man Isolationen, die dem Massengesetz nicht entsprechen. Für zusammengesetzte Strukturen aus Duralumin, Preßspanplatten oder Steinwolle ergaben sich wichtige Verbesserungen. Eine völlig elastische Befestigungsart wurde verwendet.

Summary. Different types of lightweight structures have been studied; the maximum mass was 15 kg/m², while the area of samples was about 2 m². For a simple wall the authors obtained results

differing somewhat from the mass law. For complex structures, considerable improvements were noticed; basic materials were mainly dural, compressed wood and rockwool. These structures were fixed elastically and each element independently.

1. Introduction

Le but recherché dans les études dont nous décrivons ici le commencement est de trouver des éléments légers composites qui puissent constituer soit des portes acoustiques qui donnent un bon isolement sans être trop lourdes, soit des cloisons légères amovibles sans trop de difficultés. Nous nous sommes donc fixés 15 kg/m² comme poids maximum de chacun de ces éléments, ce qui correspond à un poids maximum de 50 kg par mètre linéaire de cloison, sur toute une hauteur d'étage. Ce poids ne rend pas les éléments trop lourds au point de ne pouvoir être maniés qu'avec difficulté quand on veut déplacer une cloison.

La loi de masse appliquée à un tel élément ne donnait guère plus d'une trentaine de dB pour l'isolement à 500 Hz et une vingtaine à 100 Hz. Il fallait gagner de 15 à 20 dB sur ces dernières valeurs pour obtenir un isolement convenable par l'emploi d'éléments composites.

L'étude a donc porté sur le meilleur choix de ces éléments, leurs meilleures dispositions respectives et leur meilleur mode d'accrochage.

2. Méthode de mesure et appareillage utilisé

Les structures essayées étaient placées dans une ouverture de 2,12 m de haut, sur 0,88 m de large, dimensions voisines de celles d'une porte, mais malheureusement un peu trop petites pour un élément de cloison. Le panneau séparait 2 chambres de 60 m³ environ. Dans l'une la source était constituée par 2 haut-parleurs alimentés par un générateur B.F. à compression donnant un son de fréquence modulée à $\pm$ 35 Hz à la vitesse de 4 cycles par seconde. Un micro à bobine mobile était placé dans cette chambre près du panneau. Dans la chambre de réception un micro du même type recueillait le son et l'envoyait sur un enregistreur de niveau rapide. Un système de commutation rapide permettait de passer continuellement d'un micro à l'autre pour attaquer l'enregistreur. Les mesures étaient faites aux fréquences normalisées suivantes: 100, 125, 160 Hz et leurs doubles successifs jusqu'à 3200 Hz.

Nous ne donnerons pas ici les valeurs obtenues pour l'isolement à toutes ces fréquences, ce qui aurait alourdi l'exposé, mais seulement les valeurs moyennes pour 3 groupes de fréquences:

1) Le groupe 100, 125, 160, 200, 250, 320 Hz, que nous appellerons les basses,
2) le groupe 400, 500, 640, 800, 1000, 1250 Hz, que nous appellerons les médiums,
3) le groupe 1600, 2000, 2500, 3200 Hz, que nous appellerons les aiguës.

Au début de toute série de mesures, le micro source et le micro réception étaient placés côte à côte et les chaînes d'amplification réglées de telle sorte que les niveaux d'inscription sur l'enregistreur soient les mêmes. Comme ceci ne pouvait être obtenu pour toutes les fréquences, un terme correctif était utilisé pour chacune des fréquences ci-dessus.

3. Matériaux utilisés

Deux types de matériaux ont été essayés: *Dural* et fibres de bois compressées, type *Novopan* pour faire des cloisons soit en métal, soit en bois.

Les éléments en Novopan avaient 16 et 25 mm d'épaisseur, pesant respectivement 9,1 kg et 14,75 kg/m².

Les éléments en Dural avaient 1,5 et 2,0 mm d'épaisseur, représentant des poids de 3,9 et 5,2 kg/m².

Le mode de fixation de ces éléments est tout à fait particulier. Nous rappelons qu'un panneau léger a pour analogie électrique une self inductance représentant la masse en série avec une capacité représentant l'élasticité. Il se trouve malheureusement que ce circuit a une résonance à une fréquence située dans les basses, qui a pour valeur

$$f_0 = \frac{1}{2\pi}\sqrt{\frac{K}{m}},$$

K étant la raideur et m la masse unitaire du panneau.

Pour cette fréquence la valeur de l'isolement passe par un minimum, parfois très faible. Nous avons pensé qu'il fallait essayer de rejeter cette résonance dans le zône des fréquences les plus basses possibles en augmentant la capacité, donc en diminuant la raideur du panneau. Celui-ci a été monté le plus souple possible, en tapissant de caoutchouc l'encadrement intérieur de l'ouverture et en revêtant les bords du panneau d'un matériau très élastique, le Klegecell mou. Chaque élément tenait simplement par frottement d'une substance élastique sur l'autre.

Les isolements obtenus dans ces conditions sont bons aux basses et médiums, moins bons aux aiguës. Ce dernier affaiblissement semble dû à une étanchéité qui ne peut être parfaite, vu le mode de fixation utilisé.

Les isolements sont représentés par les courbes de la Fig. 1 pour les panneaux simples de Novopan,

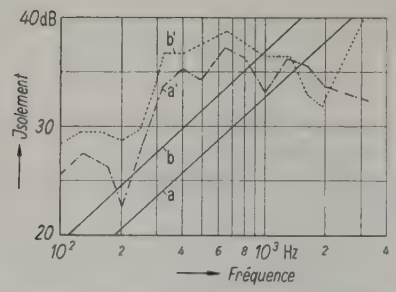


Fig. 1. Isolement théorique (loi de masse) et expérimental de deux types de panneaux simples. Son réverbérant.
(a) Novopan 16 mm, théorique;
(a') id. , expérimental;
(b) Novopan 25 mm, théorique;
(b') id. , expérimental.

et celles de la Fig. 2 pour les panneaux simples de Dural.

Les isolements qu'aurait donné la loi de masse sont représentés sur les mêmes figures par deux droites. La formule employée est celle de LONDON, puisque nous nous trouvions toujours en milieu réverbérant.

$$TL = 10 \log \frac{a^2}{\ln (1 + a^2)},$$

où

$a = \omega m / 2 \varrho c$,
m = masse par unité de surface du matériau,
ω = pulsation de fréquence,
ϱ = densité de l'air, } $\varrho c = 420$ unités MKS
c = vitesse du son. }

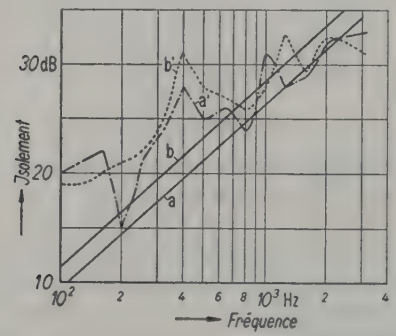


Fig. 2. Isolement de deux types de panneaux simples.
(a) Dural 1,5 mm, théorique;
(a') id. , expérimental;
(b) Dural 2,0 mm, théorique;
(b') id. , expérimental.

Nous avons volontairement donné les valeurs de l'isolement sonore normalisé:

$$D_n = L_1 - L_2 + 10 \log T/0{,}5,$$

qui correspondent aux valeurs utilisées dans la pratique entre deux bureaux normalement meublés. Notre terme correctif $10 \log T/0{,}5$ variait entre 1,4 et 1,8 dB.

4. Structures composites essayées

1) La première série d'essais porte sur les éléments métalliques. Les valeurs moyennes obtenues pour les isolements sonores normalisés sont représentées sur la Fig. 3. Il n'est jamais possible d'obtenir un bon isolement aux basses à cause de la très grande légèreté des éléments, mais il est à remarquer que l'introduction de laine de verre dans l'intervalle donne un gain d'isolement non négligeable, surtout aux médiums et aiguës.

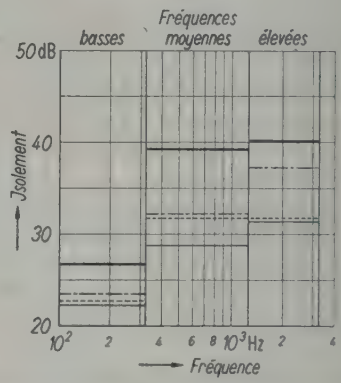


Fig. 3. Isolement moyen pour trois bandes de fréquence des panneaux de Dural seuls et couplés.
——— Dural 1,5 mm, seul (3,9 kg/m²);
- - - - - - - - - - - - Dural 2,0 mm, seul (5,2 kg/m²);
——— ——— Dural 1,5 mm et 2,0 mm à 3 cm;
——— ——— id. avec remplissage de laine de verre.

2) La deuxième série de mesures avait pour but de trouver une bonne structure en bois avec des panneaux de Novopan. Nous avons tout d'abord étudié l'influence de l'écartement des deux panneaux sur l'isolement. La distance séparant les deux éléments variaient entre 3 et 13 cm. Les résultats trouvés sont représentés sur la Fig. 4. Ils montrent que si un faible intervalle de l'ordre de 3 cm n'apporte pas beaucoup d'amélioration dans l'isolement par rapport au panneau le plus lourd dans les fréquences basses, il n'en est pas de même aux médiums et aiguës.

Il faut arriver à un intervalle de 8 cm pour avoir un gain d'isolement notable dans les basses. Si la distance dépasse cette valeur, l'isolement croit évidemment, mais au point de vue pratique, il est couteux de prendre des valeurs trop fortes dans la plupart des cas. Une porte ou une cloison amovible de plus de 12 cm d'épaisseur deviennent chères et plus difficiles à construire.

Une fois la distance entre les deux éléments fixée à 8 cm nous avons recherché quelle était la meilleure facon d'améliorer l'isolement en placant un troisième matériau dans l'intervalle.

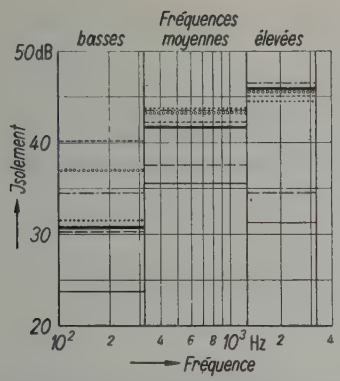


Fig. 4. Isolement moyen pour trois bandes de fréquence des panneaux de Novopan seuls et couplés avec intervalle d'air variable entre eux.
——————— Novopan 16 mm, seul (9,1 kg/m²);
— — — — — — Novopan 25 mm, seul (14,75 kg/m²);
——————— Novopan 16 mm et 25 mm à 3 cm;
++++++++++++ id. à 5,5 cm;
—·—·—·—·— id. à 8 cm;
oooooooooooooo id. à 10,5 cm;
---------- id. à 13 cm.

Ont été essayés:
— un panneau de Dural de 1,5 mm;
— de la laine de roche sous 5 cm d'épaisseur;
— un élément composé de deux panneaux de Dural de 1,5 et 2,0 mm à 3 cm l'un de l'autre;
— un tissu vinylique étanche tendu sur un cadre;
— un tissu de verre poreux également tendu sur un cadre.

Les résultats enregistrés sont indiqués par la Fig. 5. La plus forte amélioration dans les basses

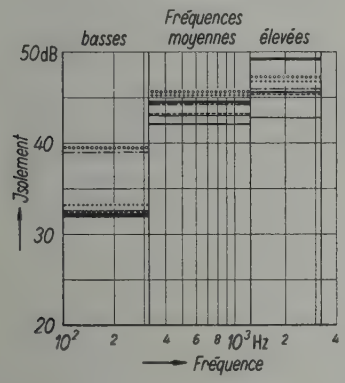


Fig. 5. Étude de deux panneaux de Novopan, un de 16 mm et un de 25 mm, à 8 cm (l'ensemble pesant 23,85 kg/m²), avec présence d'un troisième matériau dans l'intervalle.
——————— air;
---------- Dural 1,5 mm;
—·—·—·—·— laine de roche 5 cm;
——————— Dural 1,5 mm + Dural 2,0 mm;
——————— tissu vinylique;
++++++++++++ tissu de verre;
oooooooooooooo laine de roche 5 cm + tissu de verre.

était obtenue avec la laine de roche et ceci d'une facon très marquée, puisque l'isolement atteignait 40 dB en moyenne.

Résultat curieux — Le tissu de verre avait donné un gain appréciable dans la zône des médiums et des aiguës. Nous avons donc, comme dernier essai, combiné la laine de roche et le tissu de verre en les plaçant l'un contre l'autre dans l'intervalle de 8 cm laissé entre les deux panneaux de Novopan.

Les isolements sonores normalisés étaient alors de:
— 41 dB dans les basses,
— 47 dB dans les médiums,
— 49 dB dans les aiguës, valeurs tout à fait intéressantes pour un ensemble d'éléments ne pesant pas plus de 25 kg/m² au total.

3) Nous avons voulu dans une troisième série d'essais savoir ce que pouvait donner une combinaison d'élément bois et d'élément métallique. Une plaque de Dural de 1,5 mm d'épaisseur était placée à 5 ou 9 cm d'un panneau bois de 16 mm d'épaisseur, ou bien à 3 cm d'un panneau bois de 25 mm. Les valeurs trouvées sont indiquées par la Fig. 6.

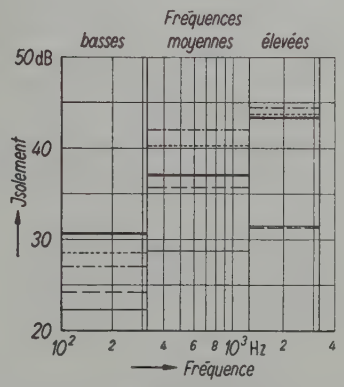


Fig. 6. Combinaison d'élément bois et d'élément métallique.
——————— Dural 1,5 mm, seul;
— — — — — — Novopan 16 mm, seul;
—·—·—·—·— Dural 1,5 mm + Novopan 16 mm à 9 cm;
---------- id. à 5 cm;
——————— Dural 1,5 mm + Novopan 25 mm à 3 cm.

L'addition d'un élément métallique améliore bien l'isolement aux aiguës, beaucoup moins aux basses et aux médiums. La combinaison qui tout en étant la meilleur marché, semble assez efficace, est celle d'un panneau Dural à 5 cm d'un panneau bois de 16 mm.

On peut ainsi espérer atteindre en moyenne:
— 28 dB aux basses,
— 40 dB aux médiums,
— 44 dB aux aiguës.

Après avoir recherché la combinaison des éléments amovibles la plus efficace, il reste à trouver le mode de fixation le plus pratique qui soit le plus voisin possible de celui que nous avions adopté pour nos essais. C'est le but de nos derniers essais en cours, qui nous l'espérons, aboutiront à trouver, surtout dans les basses, un isolement bien plus fort que celui que nous aurait donné l'application de la loi de masse.

Discussion

VOGEL, TH. M.: Je suis heureux de constater que les résultats obtenus par M. PUJOLLE sont en bon accord avec une théorie que j'ai publiée dans le Journal de Physique en 1947. Je voudrais également signaler qu'à la suite de ce travail de nombreuses vérifications expérimentales ont été faites au Laboratoire de Marseille par M. KOBRYNSKI, dont le rapport n'a malheureusement eu qu'une diffusion restreinte.

CONFÉRENCIER: Nous n'avons pas eu connaissance de ces travaux.

LANSARD, R.: Les maxima et minima des courbes d'atténuation ne sont-ils pas dus à une résonance de l'appareillage, puisqu'ils coincident pour des échantillons différents (Dural 1,5 et 2,0 mm)?

CONFÉRENCIER: C'est sans doute la fréquence propre du panneau.

GAVREAU, V.: A-t-on utilisé des matériaux autres que la laine de verre? La laine de verre présente-t-elle des avantages particuliers?

CONFÉRENCIER: Nous avons utilisé la laine de verre parce que c'est un matériau commun, facile à se procurer. Mais on obtient des résultats absolument semblables dans le cas d'autres matériaux de ce type, tels que les laines minérales, etc. . .

ZUM EINFLUSS DER SEITLICHEN BEGRENZUNG AUF DIE SCHALLDURCHLÄSSIGKEIT EINFACHER WÄNDE

von A. SCHOCH

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Zusammenfassung. Der Schalldurchgang durch eine begrenzte Platte wird theoretisch behandelt. Das Ergebnis bietet keine Erklärung der abnorm hohen Werte der Schallisolation, die bei leichten Wänden und tiefen Frequenzen gemessen werden.

Summary. The transmission of sound through a plate with finite dimensions has been treated theoretically. The result does not explain the abnormally high values of sound insulation measured on lightweight partitions at low frequencies.

Sommaire. On étudie théoriquement la transmission du son par un panneau de dimensions finies. Le résultat obtenu ne permet pas de rendre compte des valeurs anormales de l'isolement acoustique qu'on a constatées dans le cas des parois légères et des basses fréquences.

Da die empirische Schallisolation einfacher Wände mit der von CREMER [1] gegebenen Theorie des Schalldurchgangs durch eine unbegrenzte elastische Platte noch nicht befriedigend übereinzustimmen scheint — vor allem findet man auch bei sehr leichten Wänden und sehr tiefen Frequenzen Abweichungen, und zwar größere Schallisolationen als die Theorie erwarten läßt [2] — liegt die Frage nahe, ob nicht bei begrenzten Wänden Resonanzeffekte mit den Eigenschwingungen der Wände eine Rolle spielen. Qualitativ läßt sich sagen, daß die Schallisolation durch die Wandbegrenzung sicher modifiziert wird in dem Bereich von Frequenz mal Plattendicke, in dem der „Koinzidenz-Effekt" auftritt. Man kann nämlich die Schwingungen einer begrenzten Wand betrachten als Superposition der „erzwungenen" Biegewelle, wie sie auf der unbegrenzten Wand erregt würde, und von „freien" Biegewellen, die vom Rand ausgehen. Letztere geben im Koinzidenzbereich Anlaß zu zusätzlich abgestrahlten Wellen und damit zu veränderter Schallisolation [3]. Unterhalb des Koinzidenzbereichs sollte man zunächst keine Modifikation der Schallisolation gegenüber der unbegrenzten Wand erwarten, da die freien Biegewellen nicht strahlen, wenn ihre Wellenlänge kleiner als die Wellenlänge in Luft ist. Dies ist jedoch nur so lange richtig wie die Ausdehnung der Wand sehr viele Wellenlängen in Luft beträgt. Man könnte daher vermuten, daß auch die zu große Schallisolation bei sehr tiefen Frequenzen durch Mitwirkung der freien Wellen zustande kommt.

Um quantitative Aussagen über die Wirkung der Wandbegrenzungen zu ermöglichen, wurde folgendes zweidimensionales Problem behandelt: Quer in einem von zwei parallelen starren Wänden begrenzten Kanal sei eine dünne Platte der Breite a gespannt (Abb. 1). Von $z < 0$ her fallen Schallwellen ein, deren Schalldruckamplitude $p^{(1)}$ wir als Überlagerung der von RAYLEIGH eingeführten Rohrwellen-Grundformen („wave guide modes") darstellen:

$$p^{(1)} = \sum_{m=0}^{\infty} (A_m^{(1)} e^{ik_m z} + B_m e^{-ik_m z}) \times$$

$$\times \varepsilon_m \sqrt{\frac{2}{a}} \begin{matrix} \cos(m\pi x/a) \\ \sin(m\pi x/a) \end{matrix} \text{ für } \begin{matrix} m = 0, 2, 4 \ldots \\ m = 1, 3, 5 \ldots \end{matrix}, \quad (1)$$

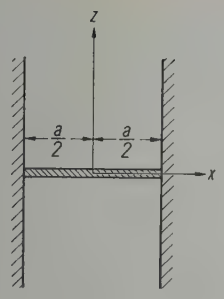


Abb. 1. Kanal mit Querwand.

wo $A_m^{(1)}$, B_m die Amplituden der in $+z$- bzw. $-z$-Richtung laufenden Rohrwellen sind;

$$\varepsilon_0 = 2^{-1/2},\ \varepsilon_m = 1 \quad \text{für} \quad m > 0$$

und $$k_m = k\left[1 - \left(\frac{m\pi}{ka}\right)^2\right]^{1/2},\ k = \frac{\omega}{c}$$

mit ω = Frequenz,
c = Schallgeschwindigkeit in Luft.

Jede der Teilwellen in (1) besteht aus zwei ebenen Wellen, deren Fortschreitrichtungen einen Winkel ϑ_m mit der z-Achse bilden, der durch

$$\sin\vartheta_m = \frac{m\pi}{ka}$$

gegeben ist, so daß also auch $k_m = k\cos\vartheta_m$. Für $z > 0$ sei der Kanal reflexionsfrei abgeschlossen, so daß die durchgelassenen Wellen dargestellt werden durch

$$p^{(2)} = \sum_{m=0}^{\infty} A_m^{(2)}\, e^{ik_m z}\, \varepsilon_m \sqrt{\frac{2}{a}} \begin{matrix}\cos(m\pi x/a)\\ \sin(m\pi x/a)\end{matrix}. \qquad (2)$$

Die durchgelassenen Amplituden $A_m^{(2)}$ und die reflektierten Amplituden B_m sind mit den einfallenden $A_n^{(1)}$ verknüpft durch die Bewegungsgleichung der Platte und die Bedingung, daß die Schnelle der Platte auf beiden Seiten mit der z-Komponente der angrenzenden Luft übereinstimmt. Die Verknüpfung führt auf ein System von linearen Gleichungen für die unendlich vielen $A_m^{(2)}$, wenn man die $A_n^{(1)}$ als gegeben betrachtet. Es ist bemerkenswert, daß dieses Gleichungssystem explizit lösbar ist.

Wie zu erwarten, regen gerade n nur gerade m an, und ungerade nur ungerade. Für die von einem einzigen $A_n^{(1)}$ angeregten „durchgelassenen" $A_m^{(2)}$ ergibt sich für gerade m, n (für ungerade m, n erhält man ein analoges Ergebnis) [1])

$$\frac{A_m^{(2)}}{A_n^{(1)}} = D_m\left\{\delta_{mn} + \varepsilon_m\varepsilon_n\cos\frac{m\pi}{2}\cos\frac{n\pi}{2}\cdot i\sigma\Omega\cos\vartheta_n\cdot D_n\times\right.$$
$$\left.\times\left[\frac{F(1+\Omega\sin^2\vartheta_m)(1+\Omega\sin^2\vartheta_n) + G(1-\Omega\sin^2\vartheta_m)(1-\Omega^2\sin^2\vartheta_n) + 2H(1-\Omega^2\sin^2\vartheta_m\sin^2\vartheta_n)}{FG - H^2}\right]\right\}. \qquad (3)$$

Hierin sind δ_{mn} das Kroneckersymbol,

$$D_m = [1 - i\sigma\Omega\cos\vartheta_m(1 - \Omega^2\sin^4\vartheta_m)]^{-1}$$

der Cremersche Ausdruck für die durchgelassene Amplitude, $\Omega = \omega/\omega_{krit}$ eine dimensionslose Frequenz (bei $\omega = \omega_{krit}$ ist die Biegewellengeschwindigkeit $c_B = c$), und die dimensionslose Größe

$$\sigma = \frac{\sqrt{12}\,\varrho_P\, c}{2\,\varrho c^D}$$

enthält alle Materialkonstanten (ϱ_P = Dichte, c_D = Geschwindigkeit der Dehnungswellen in der Platte).

F, G, H sind Funktionen von Ω und $\zeta = \frac{\varkappa a}{2\pi} = \frac{1}{2\pi}\frac{\omega}{c_B}a$, und zwar ist

$$F = -i\mu + \pi\zeta\,\mathrm{ctgh}\,\pi\zeta - \sum_{n=-\infty}^{+\infty} D_n\zeta^2(\zeta^2 + n^2)^{-1} + H$$

$$G = -i\mu + \pi\zeta\,\mathrm{ctg}\,\pi\zeta - \sum_{n=-\infty}^{+\infty} D_n\zeta^2(\zeta^2 - n^2)^{-1} + H$$

$$H = \tfrac{1}{2}\sum_{n=-\infty}^{+\infty} D_n.$$

Die Größe μ hängt von den Randbedingungen für die Platte ab. Es ist $\mu = 0$ im Fall der am Rande „gelenkig gelagerten" Platte, $\mu = \pi\zeta$ im Fall einer Biegewellen nicht reflektierenden Randbefestigung. Diese Randbedingungen entsprechen den Extremfällen einer dissipationsfreien und einer vollkommen dissipierenden Randbefestigung.

Das reichlich komplizierte Ergebnis (3) zeigt, daß im durchgelassenen Schall im allgemeinen alle Wellentypen vertreten sind, auch wenn die einfallende Welle nur *eine* Richtung ϑ_n enthält. Offenbar reduziert sich (3) auf die Cremersche Formel, wenn die eckige Klammer verschwindet.

Eine Anwendung wurde von dem Ergebnis vorläufig nur gemacht, um nachzuprüfen, ob es vielleicht die von London festgestellten Unstimmigkeiten bei sehr kleinen Werten von Ω zu erklären vermag. Dazu wurde für eine Aluminiumplatte und die Werte $\Omega = 0{,}004$, $\varkappa a/2\pi = 5$ bis 6 [2]) das Verhältnis von einfallender zu durchgelassener Energie (die „Durchlässigkeit", im vorliegenden Fall gleich $|A_0^{(2)}/A_0^{(1)}|^2$) in Abhängigkeit von $\varkappa a/2\pi$ berechnet (Abb. 2). Die Durchlässigkeit durchläuft in der Umgebung der Resonanz mit einer Biegungseigenschwingung alle Werte von 0 bis 1; im Mittel beträgt aber die Abweichung von der Durchlässigkeit

[1]) Die Wiedergabe der Rechnung muß einer besonderen Darstellung vorbehalten bleiben.

[2]) $\varkappa a/2\pi$ ist die in Biegewellenlängen gemessene Seitenlänge a. Die Werte entsprechen z.B. einer Platte von 0,4 mm Dicke und 1 m Seitenlänge bei einer Frequenz von 100 Hz.

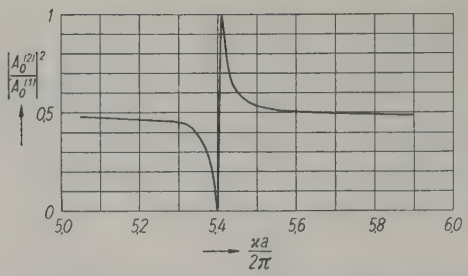


Abb. 2. Durchlässigkeit einer Platte aus Aluminium für $\Omega = 0{,}004$, in Abhängigkeit von der in Biegewellenlängen gemessenen Seitenlänge.

der unbegrenzten Platte (hier gleich 0,5) höchstens einige Prozent. Abb. 2 bezieht sich auf eine dissipationsfreie Randbefestigung. Im Fall des reflexionsfreien Randes ist die Resonanzstelle praktisch verschwunden und die Durchlässigkeit weicht nirgends nennenswert von der der unbegrenzten Platte ab.

Dieses Ergebnis läßt Zweifel an der Realität der hohen Schallisolationswerte im Bereich $\omega \ll \omega_{krit}$ berechtigt erscheinen.

Schrifttum

[1] CREMER, L., Akust. Z. 7 [1942], 81.
[2] LONDON, A., J. Res. nat. Bur. Stand. Wash. 42 [1949], 605.
[3] SCHOCH, A. und FEHÉR, K., Acustica 2 [1952], 189.

Diskussion

JUNGER, M. C.: (1) The scattering action of elastic scatterers shows strong frequency dependence and resonance effects in the vicinity of natural frequencies of the scatterer. (2) For cylindrical radiators the cut-off frequency, where $k_m = 0$, may coincide with very large reactive impedances, thus eliminating the corresponding mode of vibration; the action of the fluid then resembles that of a dynamic vibration damper. Does this hold for plane radiators, such as an elastic plate? Does the plate act as though it were rigid in that mode?

VORTRAGENDER: (1) The same is the case with elastic plates in the vicinity of natural frequencies, see Fig. 2 of the paper. (2) Except in the vicinity of resonance frequencies, there is no indication of a considerable modification of the impedance of the plate by reaction of the air. Modes of high reactance ($k_m = 0$) play only a subordinate part in transmission, because they are excited only weakly.

KUHL, W.: Ist die gemessene zu hohe Schalldämmung bei tiefen Frequenzen, bei denen die beiden Meßräume nur wenige Eigenfrequenzen in dem angeregten Frequenzbereich (z.B. 1/3 Oktave) haben, dadurch zu erklären, daß im Senderaum großer Schalldruck nur bei Eigenfrequenzen dieses Raums auftritt, daß aber die durch die Trennwand hindurchtretende Schallenergie im Empfangsraum, mit anderen Eigenfrequenzen, keinen großen Druck erzeugt?

VORTRAGENDER: Dieser Effekt sollte sich dadurch ausgleichen, daß ebenso häufig der Empfangsraum in Resonanz angeregt wird mit entsprechend überhöhtem Schalldruck. Vielleicht ergibt sich aus dem Vortrag von Herrn PEUTZ ein Hinweis auf das Verhalten einer Wand, wenn die Raumeigenfrequenzen nicht mehr dicht liegen.

PEUTZ, V. M. A.: I agree with Dr. SCHOCH that the relatively high value of the transmission loss at low frequencies cannot be explained by the effect of the natural frequencies of the transmitting and/or the receiving room.
I would like to make another remark concerning Dr. SCHOCH's theory. In order to study the influence of the mounting of the samples we have made some measurements of the transmission loss at normal incidence of a number of foils (copper, lead, rubber) in a tube of "infinite" length. Near the resonance frequency the transmission loss of a red copper foil showed the same frequency-dependence as the theory of Dr. SCHOCH predicts.

CREMER, L.: Die Theorie der unendlichen Platte gibt eine höhere Isolation als es der Masse zukommen würde, wenn die Steifigkeit die Masse weit überkompensiert, also wenn die kritische Frequenz tief liegt. Das Hinzukommen der freien Wellen würde diese gute Wirkung teilweise zunichte machen, da sie oberhalb der kritischen Frequenz gut abgestrahlt werden. Falls es aber gelänge, die Biegewellen bei der Ausbreitung oder am Rande zu dämpfen, wäre trotzdem die Möglichkeit gegeben, leichte und doch gut isolierende Wände zu schaffen.

TRANSMISSION OF REVERBERANT SOUND THROUGH WALLS

by R. V. WATERHOUSE

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Summary. Further computations of LONDON's expression for the transmission loss *TL* of double walls revealed significant differences between the computed results and experimental data, both at discrete frequencies and as regards the functional dependence of the loss on frequency. To establish more precisely this frequency dependence, particularly in the region of the "coincidence effect", more refined measurements were made giving *TL* as a continuous function of frequency. Some results are presented.

Sommaire. En poursuivant le calcul numérique de l'expression de LONDON relative à l'isolement sonore des parois doubles, on avait trouvé des différences notables entre les résultats de ce calcul et les résultats expérimentaux, tant pour certaines fréquences que pour la variation elle-même de l'isolement avec la fréquence. Afin de pouvoir déterminer cette variation avec plus de précision, surtout dans la région de l'"effet de coïncidence", on a procédé à des mesures plus systématiques, donnant cette variation d'une manière continue. On indique dans cet article quelques uns des résultats ainsi obtenus.

Zusammenfassung. Weitere Berechnungen zu LONDON's Formel für die Schalldämmzahl von Doppelwänden ergaben deutliche Abweichungen zwischen den errechneten Werten und den experi-

mentellen Ergebnissen, sowohl bei einzelnen Frequenzen, als auch hinsichtlich des Verlaufs der Schalldämmung mit der Frequenz. Um diese Frequenzabhängigkeit exakter festzulegen, besonders im Gebiet des „Koinzidenzeffektes", wurden sorgfältige Messungen durchgeführt, welche die Schalldämmzahl als kontinuierliche Funktion der Frequenz ergaben. Einige Ergebnisse werden angegeben.

1. Introduction

Some further computations have been made of LONDON's formula for the transmission loss TL of double walls [1].

LONDON's theoretical treatment differs from others in the way that the dissipation factor is introduced. The impedance per unit area of each panel of the double wall is taken as

$$Z = 2\,R/\cos\theta + \mathrm{j}\,\omega m,$$

where m is the mass per unit area of the panel, R is a dissipation factor, θ is the angle of incidence.

This factor R was introduced in the first place to account for insulation values higher than the mass law predicted at low frequencies. CREMER's theory of the coincidence effect was incorporated in the above treatment, except that, instead of using a complex stiffness to account for dissipation, the factor R was used as before.

Thus in effect one dissipation factor R was used to take into account all losses, compressional, flexural and by radiation, and in addition the losses occurring at the edges of the panels. This procedure appeared to be justified empirically in the case of single panels of aluminium, plywood and plasterboard, where it gave good agreement with experimental results. However, the values of R required to give agreement were considerably larger than the known attenuation figures for compressional and flexural waves in the materials.

In applying the above treatment to double plywood panels and using the same value of R as had been used for the single wall cases, it was found that the computed results did not agree well with experiment, particularly at frequencies above the coincidence frequency. In order to make the theory agree with experiment, the factor R was made frequency-dependent. This empirical step gave reasonable agreement for both single and double plywood walls [1]. It was possible to vary R and adjust the double wall curve without altering much the single wall curve, since the former was far more sensitive to changes of R than the latter.

2. Results of computations

Further computations have now been made which enable the theory to be compared with experiment for some double plasterboard walls. Figures 1 and 2 show these results for 12.5 mm and 25 mm plasterboard, respectively, where the resistance factor R has been kept the same for all frequencies.

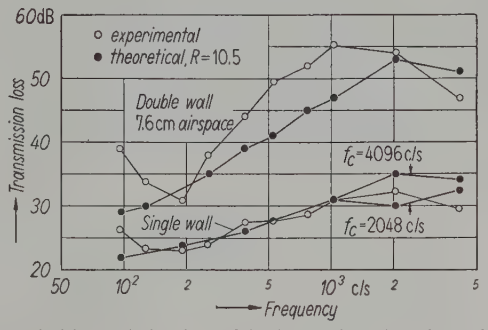


Fig. 1. Transmission loss of 12.5 mm plasterboard panels.

It is apparent that agreement is not adequate under this condition of R being constant. However, these computed values are of interest as showing the general shape of the transmission loss curve predicted for double walls, under the assumptions stated. It is seen that the coincidence effect gives only a small dip in the theoretical curve, smaller than that appearing in the experimental curve.

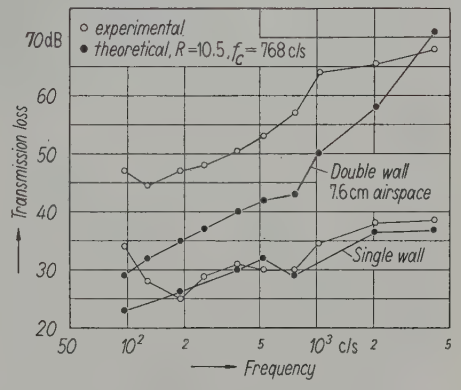


Fig. 2. Transmission loss of 25 mm plasterboard panels.

The agreement between theory and experiment could certainly be improved by making R frequency-dependent, as was done in the case of the double plywood panels [1]. However, here the single wall curves agree fairly well, and altering R to make the double wall curves agree better would to some extent upset the agreement of the single wall curves.

The conclusions drawn from these results were that R had to be made frequency-dependent to enable the double wall theory to match experiment and that since R then became the dominant factor in the TL formula, it was desirable to re-examine the question of the dissipation and to account for it in a less arbitrary fashion.

3. Results of measurements of *TL*

In attacking this problem, it was decided as a first step to establish more accurately the shape of the *TL* versus frequency curve, particularly in the region of the coincidence effect. Accordingly, measurements were made of the *TL* of some of the panels considered above, taking advantage of a recently developed method for recording the *TL* directly as a continuous function of frequency. This method will be described in a paper elsewhere.

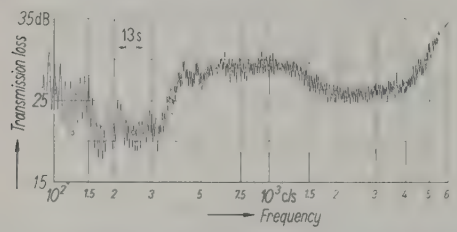


Fig. 3. Transmission loss of single 12.5 mm plywood panel.
o = Experimental points measured by another method.

Some of these new measurements are given in Fig. 3, 4, 5 and 6. The curves do not give the exact *TL*, as the correction for the absorption of the quiet room is not included. This correction amounts to about — 2 dB at the higher frequencies and is small enough to be ignored in considering the main features of the *TL* curves.

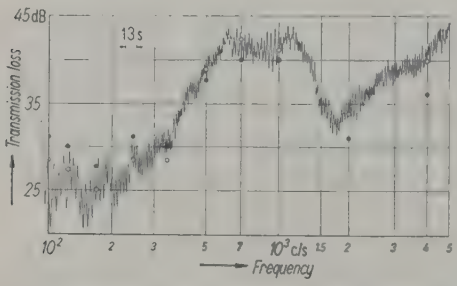


Fig. 4. Transmission loss of double 12.5 mm plywood panel, 76 mm airspace.
o = Experimental points measured by another method.
• = Above points corrected for quiet room absorption and microphone calibration.

The new data agree fairly well with earlier data and are considerably more revealing. Firstly, the continuous records enable an average *TL* figure to be taken that is in general more accurate than could be obtained by measuring with one frequency band centered on the nominal frequency. Secondly, the sharp drop in the *TL* curve in the coincidence region is shown much more clearly in the continuous record than before. The drop is sharper for the plasterboard than for the plywood and is also greater. The general shape of the curves for the plasterboard is reminiscent of those that result from considering the panels to be pure masses, coupled by an elastic airspace, the sound field being random; however, the large dips in the curves cannot be accounted for in this way either as regards their magnitude or the frequencies at which they occur and there seems little doubt that they are caused by the coincidence effect.

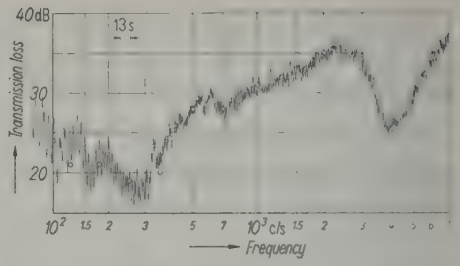


Fig. 5. Transmission loss of single 12.5 mm plasterboard panel.
o = Experimental points measured by another method.

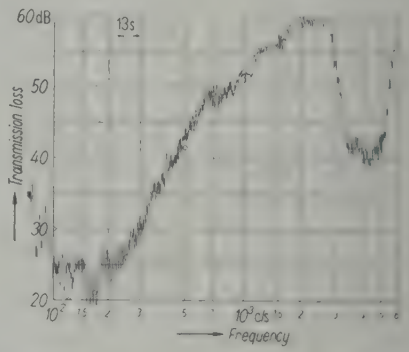


Fig. 6. Transmission loss of double 12.5 mm plasterboard panel, 76 mm airspace.
o = Experimental points measured by another method.

From the known elastic constants of the materials, the coincidence frequencies can be computed and they check with the experimental data.

It seems probable that the correct theoretical explanation of these *TL* curves rests with the proper introduction of the dissipation factor into the theory. While CREMER's treatment does not seem sufficient to account for the losses and the shape of the *TL* curve at frequencies below the coincidence effect, LONDON's treatment does not adequately account for the high frequency part of the curve. It is hoped to find another way of introducing the loss factors, perhaps to some extent a synthesis of both the above methods, which will account for the facts.

Reference

[1] LONDON, A., J. Res. nat. Bur. Stand. 42 [1949], 605.

Discussion

KORN, T.: What was done to diffuse the sound in the transmission rooms?

LECTURER: In the source room ten loudspeakers were used, giving a warble tone. Both rooms were quite reverberant, the walls being left bare concrete. Six randomly spaced pressure microphones were used in each room, and thermocouples were used for rectifying.

PEUTZ, V. M. A.: What are the dimensions of the panels and the rooms used?

LECTURER: Dimension of panels: $1.83 \times 2.29\ m^2$, quiet room: $2.67 \times 3.66 \times 4.88\ m^3$, loud room: $2.97 \times 2.90 \times 3.64\ m^3$.

JORDAN, V. L.: Did the speed of recording affect the amplitude of the irregularities on the transmission loss versus frequency curve?

LECTURER: The experiment showed that it did not.

CLOSING LECTURE

by H. B. G. Casimir
Philips Research Laboratories, Eindhoven, Netherlands



Art, Science and Technology

In the reaction of man towards the world around him we can distinguish two different tendencies. There is the *engineer*, the man who tries to dominate the forces of nature, to protect himself against them if necessary, to use them if possible, the man who tries to form matter at his will. This is not pure materialism and the moving force is not only the desire to improve the lot of mankind: the creative urge is deeply rooted in the human race and there enters a spiritual factor in every technical achievement because it shows the victory of the intellect over the material world. But there is also the *physicist*, the natural philosopher, who tries to penetrate into the nature of things, to find relations between the apparently unrelated, the man that is driven by a profound curiosity and who looks for harmony in a consistent picture of the physical world. These two tendencies are not always at peace. The conflict between them may even exist in our own hearts, for the man of action will know moments when he longs for quiet meditation whereas the pure scientist feels at moments unsatisfied with his own efforts because he has created nothing "real". And if the conflict exists inside ourselves it can be even more pronounced between exponents of the two ways of thinking. Yet it is obvious that our civilization of today is the result of a close cooperation of the scientist and the engineer: their joining of forces is perhaps the most characteristic feature of our industrial and technical age.

A few historical remarks may help to illustrate the point. The Greeks had both superb craftsmanship and profound philosophy. But the knowledge of the craftsman was mainly empirical and hardly influenced by the systems of thought of the great philosophers, although there were notable exceptions. Again the philosophy of St Thomas Aquinas did not help the architects of the cathedrals to solve their constructional problems. It is often said that the Renaissance was the beginning of a new era in natural philosophy, an era characterized by systematic experimentation and by the application of quantitative mathematical analysis to the interpretation of experimental data. This is true, but it is also true that it marked the beginning of the application of the results of exact science to the solution of engineering problems.

Christiaan Huygens is a good example. He was certainly a great physicist who made notable contributions to dynamics, optics and astronomy but he was also one of the first great technical physicists. When he developed the theory of geometrical optics it was because he wished to construct better telescopes. Much of his work on dynamics was done in connection with his studies on clocks and, as a matter of fact, the way in which he tackled the problem of constructing a reliable timepiece was very analogous to the approach of a modern engineer — the difference being of course that Huygens could not take the formulae from a textbook but had to derive them from scratch.

It seems hardly necessary further to elaborate on these historical developments. Nor do I want to speak today on the problem of reconciling technology and fundamental science, although it is the key to successful industrial research and development.

What I should like to emphasize today is that, whereas a union — be it at times a somewhat tumultuous one — has been achieved between technology and science, there has at the same time been created a schism between art and technology, between the artist and the artisan.

In ancient Greece, in medieval Europe, there was hardly a distinction between the two. The masons and stonecarvers of the middle ages produced works of great beauty, but they were also workmen taking part in the erection of buildings that are impressive even by the standards of modern technical constructions. Benvenuto Cellini was an artist, but he was also struggling with the technological problems of what today is known as precision casting. These things have changed and although every form of art requires a certain technology, yet this technology has on the whole not followed the development of our industrial age. Bridging the gap between abstract science and craftsmanship has created our civilization but it has also destroyed the unity between arts and crafts. The artist regards our industrial civilization with understandable distrust. And not only the artist: for all of us there is a discrepancy between intuitive feeling and technical knowledge and habits. This is clearly revealed by the use of metaphors in poetic language and even in daily speech. A metaphor based on older crafts is alright but more modern expressions are simply ridiculous. It is more or less acceptable to say that a character is like iron, forged by the blows of fate, but much less acceptable to say that it has been shotpeened by the course of events. Yet everyone familiar with the technique of shotpeening, where a surface is rendered more resistant by the impact of numerous small particles, will have to admit that this is an apt comparison. In the streets of our towns there are rather more bicycles than buffaloes, but I distinctly remember at least one Dutch poem where gathering clouds are compared to a herd of buffaloes. There are very few poems mentioning bicycles. Similarly there is a distrust of modern apparatus: musical instruments involving amplifiers and other electronic gear have not been wholeheartedly accepted by serious musicians; not only because of the shortcomings of such instruments — they might be improved — but because of the very circumstance that they use devices with which our feelings are not yet sufficiently familiar. Compare this to the great Italian violin builders who, of course used the best technical means available at the time.

This schism between art and technology, between our feeling and our daily experience is one of the great problems of our age.

What has all this to do with electro-acoustics? Whereas in other branches of science the physicist and engineer can do their work without paying heed to the relations between art and science, the acoustician is put face to face with this issue and has to attempt a solution. For although neither the physics of sound nor the techniques of electro-acoustics can be regarded as finished subjects, yet it seems to me the problems of electro-acoustics are today, to a large degree, artistic and psychological. The tools put at our disposal by modern science and modern technology are getting more and more perfect, and the problem of how to use these tools for the reproduction and recording of music and perhaps for the creation of novel musical instruments cannot be solved without consideration of artistic values. The task is not an easy one. In moments of leisure the scientist may play the violin, enjoy a concert and be on friendly terms with the professional artist, but to achieve a true synthesis is quite a different thing and much more difficult. The scientist has to renounce his all too frequent conceit and to recognize that his knowledge of frequency spectra and of the dynamics of vibrations does not explain the essential beauty of a musical tone or the spiritual content of a composition. The artist on the other hand has to conquer his distrust and to realize that, although the sounds he produces can be analyzed by physical science and modified at will by processes of recording and reproduction, this does not infringe his rights as an artist.

Acoustics has a noble tradition. In the culture of the Greeks it was an example that seems to disprove my earlier remarks, for here philosophy, quantitative observation and art worked closely together. It is essential for the development of acoustics in general, and electro-acoustics in particular, again to bridge the gap and to achieve unity. In so doing acoustics will not only assure its own future but also make a momentous contribution to the solution of one of our most fundamental problems.

EXCURSIONS

REPORT OF THE EXCURSIONS HELD DURING THE CONGRESS PERIOD FOR MEMBERS AND ASSOCIATES

VISIT TO THE BROADCASTING STUDIOS AT HILVERSUM, ON FRIDAY JUNE 19TH, 1953.

Concurrent excursions were made to the Technical Department of the "Nederlandse Radio-Unie" and to the studios of the Broadcasting Corporations A.V.R.O., N.C.R.V. and V.A.R.A.

In the studio buildings several recording and control rooms were shown, having newly designed amplifier systems.

A drama studio, incorporating many sound effects obtained by electrical and acoustical means, gave an example of the most complicated electro-acoustic apparatus now in use.

In the AUDIO FREQUENCY LABORATORY an exhibition was arranged of specimens of standard amplifiers, power supply and auxiliary units, such as are being applied in today's N.R.U. studio equipment. All types of units are constructed in chassis of uniform shape and size, to enable the exchange of units to be made in a few seconds without the use of tools.

The principal units are a 56 dB, 300 mW mike-to-line amplifier (two stages) and a 10 W power amplifier. The applied amount of feedback (over 40 dB) gave excellent properties to the amplifiers; e.g. distortion figures of 0.25 % max., A.F. response within $\pm$ 0.5 dB, and correct transient response.

Other important units shown were a standard modulation peak-level indicator, giving correct indication on a quasi-logarithmic scale for pulses of 10 ms and longer, a standard oscillator-amplifier for magnetic recording and a standard magnetic tape playback amplifier, employing a pre-emphasis of 35 μs according to C.C.I.R. recommendations. The whole system was developed, electrically and mechanically, by the N.R.U. Engineering Dept.

In the SOUND RECORDING LABORATORY were shown amongst others things

a test stand for magnetic tape, enabling one to measure all important parameters of a magnetic tape, both electrical and mechanical;

an electrodynamic cutter for fine-groove recording, having feed-back from the needle point velocity to the amplifier input, thus giving extremely low distortion;

a disc recording machine of Dutch manufacture (Strefofoon Ltd., Rotterdam), in which the turntable slides, instead of the cutter bracket. Starting and stopping of the recording process is done by a pushbutton controlled servo mechanism. This machine has proved to be almost as easy to handle as a magnetic tape recorder and to run for several thousands of hours without any maintenance;

a professional playback turntable by the same manufacturer for 78, 45, and $33\frac{1}{3}$ r.p.m., containing three different synchronous motors and worm gears. One of the speeds is chosen by switching the corresponding motor. Only the (fastest) running worm-wheel is coupled to the turntable shaft by a friction freewheel coupling;

a demonstration of measuring *wow and flutter* of a playback turntable, using a magnetic recording disc. This method enables the exclusion of errors, due to excentricity, because this disc is played back in the same position on the turntable as it was recorded just before.

DEMONSTRATIONS OF AN OUTSIDE BROADCAST VAN

A van was demonstrated, having not only normal audiofrequency facilities, but in addition two walkie-talkies of full broadcast quality. With a power of only 0.2 W, a carrier frequency of approx. 100 Mc/s frequency-modulated, a working range of a few hundred meters was guaranteed.

In the van, a transmitter of 10 W, amplitude modulated on a carrier frequency of 45 Mc/s, was used to enable talk-back.

In addition smaller F.M. transmitters were available of pocket size; for special cases this provides the possibility for the commentator to work under unobserved conditions.

Demonstrations in the acoustical laboratory

Automatic reverberation time measuring equipment

An apparatus has been developed, recording automatically reverberation-time versus frequency. About 300 measurements covering the frequency-range from 35 to 10 000 c/s, are plotted in less then 20 minutes. The frequency intervals for the low frequencies are 1 to 2 c/s. Room and other resonances are easily detected.

Magnetic reverberation machine

A machine, using an endless magnetic tape and only *one* reproducing head, may give a sufficient number of reflections to ensure a good simulation of reverberation. By means of a short delay-time and a stable feed-back circuit, a reverberation-time of 3 s was obtained; to avoid the effect of "pail-sound", it was necessary to have a nearly flat overall frequency-characteristic. This was achieved by adding the original signal in opposite-phase and predetermined amount to the series of reflections.

Transmissions from concerthalls, having a short reverberation time, can be improved by inserting this apparatus in one or more microphone circuits.

Small reverberation box

For the same purpose a small reverberation box has been constructed. The box, having dimensions of $1.70 \times 1.40 \times 1.05$ m^3, is made of plywood of 25 mm thickness. The inner walls are covered with glass plates. A second set of glass plates is attached flexibly to the walls, forming an inner box. The reverberationtime obtained is 2 s at 1500 c/s and nearly 1 s at 10 000 c/s.

Selective sound level meter

A sound level meter has been made with 18 third octave band-pass filters, according to the proposals of the "Provisional code for measurements of sound insulation" *), and is being used for the measurement of impact sound transmissions and airborne sound insulations. The electrical noise level of the apparatus itself is very low, the equivalent resistance being only 30 Ω at the 200 Ω winding of the input transformer. This corresponds to a sound pressure level of 2 dB for the 100 c/s filter with a normal sensitive microphone.

White noise generator

A small noise generator for the measurement of sound insulation is based on the super-regenerative principle. It contains a super-regenerative detector, which amplifies the electronic noise of the tube itself. The external dimensions are $21 \times 16 \times 16$ cm^3.

Condenser microphone with cardioid directional response

A cardioid condenser microphone has been developed with a flat frequency response up to 13 000 c/s ($\pm$ 2 dB). It contains a nickel and an insulating triafol membrane. The nickel membrane, which is very stable, is made by electro-plating chromium steelplate; the nickel foil can easily be removed from the steel plate.

Measuring apparatus for membrane stress of condenser microphones

For the research on the stability of several kinds of membrane materials an apparatus has been made by which the stress of the membrane at the complete microphone can be measured. A d.c. voltage is supplied to the microphone and the increase of the capacity is used to measure the stress, which can be directly read from a dial in dyne/cm^2.

Recording of polar diagrams of microphones in an anechoic room

The microphone to be measured is mounted on a turntable, placed in an anechoic room. The turntable moves synchronously with a polar diagram recorder, allocated in a separate measuring room. The walls of the anechoic room are covered with about 4200 wedges, made of "Estanisol", a product of tin slagwool. The length of the wedges is 50 cm, the cut-off frequency is 160 c/s.

After the excursions a reception was given by the Municipality of Hilversum, in the form of a cocktail party, allowing an inspection of the interior of the Town Hall, famous for its modern architecture by DUDOK.

An evening concert, played by the "Radio Philharmonisch Orkest", in the studio of the Broadcasting Corporation K.R.O., concluded the excursion to the Dutch Radio-City of Hilversum.

*) PARKIN, P. H., Report of the 1948 Summer Symposium of the Acoustics Group, The Physical Society, London 1949, p. 36.

VISIT TO N.V. PHILIPS' PHONOGRAPHISCHE INDUSTRIE, BAARN, ON FRIDAY JUNE 19TH, 1953.

General

The visit to "Philips' Phonografische Industrie" in Baarn was meant to give the members and their guests some idea of the activities of Philips in regard to gramophone records.

The acetate production (the recording of the master discs) is not described, as the main purpose of the visit was to look at the fabrication and mass production methods rather than the electro-acoustical techniques. In the following account, it is there-

fore assumed that the factory receives a certain number of recorded acetates and that it starts to manufacture records from these discs. The successive stages of production will now be briefly given.

Galvanic and chemical processes

As the recorded acetate is very fragile and easily damaged, a copy is first made which is mechanically stronger and less sensitive to damage. For this purpose the acetate is chemically silver plated by a method similar to that used for the silver plating of mirrors. The silvered acetate is then copper plated in an electroplating bath. The extremely thin silver film on the acetate is thus reinforced by a thick copper layer. By means of a special technique the acetate is then separated from the copper disc. It appears that the silver film adheres strongly to the copper and detaches completely from the acetate. The copper disc with silver face is called the "negative", or colloquially, the "father". When a father has been made, the acetate plays no further part in the fabrication process and is stored in the files. If necessary it is usually possible to repeat the afore mentioned operation if another father is required, but sometimes pieces of lacquer, adhering too strongly to the silver film, are torn off the acetate when separating it from the father. If this happens the acetate is then worthless, and a new one may have to be made.

Since the acetate original has grooves, the negative or father has upstanding ridges on its metallic surface. This means that a father cannot be played back with a pick-up, so that a simple playback test to check whether the print conforms to the recording on the acetate is not possible. It should be noted that the father is uniquely valuable, since it is the first exact print of the original recording, and handling should be reduced as much as possible. Of course one would be able to mould an actual record direct from the just finished father or negative. During the moulding process, the negative would then press grooves into the record material which are an exact facsimile of the ridges on the negative. The record thus produced could then be played back in the usual way. This process, however, is only applied in very special cases where the number of records required is small, because, if the negative is damaged, the recording needs to be made again, unless another negative can be made from the acetate original. If the father is to be spared copies of the father must be made along other lines.

Exact copies of the father are obtained through an intermediate stage. A positive or "mother" is made from the father in an electroplating bath in roughly the same way as the father was prepared from the acetate. This positive will be identical to the acetate and can therefore be played back with a pick-up. By playing back and inspecting this mother by means of a microscope, any irregularities which might be present in the grooves can be detected.

Along exactly the same lines a "son" or "stamper" is made from the mother, which of course will be a facsimile of the first metal copy, being itself also a negative. This son or stamper is now ready for the process of moulding the gramophone records.

It is obvious that by following this method, all practical precautions have been taken to prevent the loss of a recording: it is now possible to make several positives from the first negative, and a large number of stampers from the positives with less risk of damaging one of the metal parts.

Preparation of the raw material

As the regular output of the record manufacturing plant of PPI consists partly of normal shellac records and partly of unbreakable p.v.c.-type*) records, two types of raw material have to be prepared before the actual pressing process can start. The shellac compound is prepared in a special ball mill by milling all constituents to a very fine grain, dry-mixing them in exact proportions and plasticising them in a special mixer. After preparing a certain amount of this material it is scraped off the heated cylinder and fed into a calender in which the material is flattened out to the exact thickness and cut into "biscuits" of a given dimension and weight.

The production of the material for the unbreakable plastic records is similar to that for the shellac records as far as the dry-mixing is concerned. However, the compounding takes place not in mixing rolls but in an extrusion mixer. This machine gives the finished raw material in the form of chips or pellets which in certain respects is advantageous for this material.

The pressing process

In the moulding of records from stampers, it is most important that the record should be perfectly flat and accurately concentric (to prevent the phenomenon of "wow"). Accordingly the stampers are "centred" optically and turned smooth at the back by means of a special turning lathe.

Two stampers are mounted in the press by clamping rings and are heated up. A ball of preheated record material, either of the shellac or of the p.v.c.-type, is put between both stampers and the press is closed under a very high pressure. The material is moulded on both sides by both stampers at the same time, the ridges of the stampers forming grooves in the material. When the stampers

*) Polyvinylchloride, known under trade names such as Vinelyte.

have cooled down, the press can be opened and the record can be removed. After the pressing the edge of the record is polished and every record is submitted to a visual inspection. Furthermore there is an auditory inspection of a large number of records, which are picked out at random as the production comes off the press. Only in this way is one able to check the quality of the actual product during the production process. After these inspections the records are put into their covers and stocked.

VISIT TO THE N.V. PHILIPS' GLOEILAMPENFABRIEKEN, EINDHOVEN, ON MONDAY JUNE 22ND, 1953.

This day's programme consisted of:

a) a demonstration of sound reproduction in the studio of the Main Industrial Division of Electro-Acoustics (ELA),

b) a visit to a number of production departments,

c) a number of demonstrations in the Research Laboratories.

Short descriptions of the last-named demonstrations are given below. References to Philips publications are given. Many of the results shown to the Congress have not yet been published in which cases references to earlier work on the same subjects are given.

Where the "Philips technical Review" is quoted the same articles will be found in the French, German and Dutch editions, but on slightly different pages.

X-ray image intensifier (M. C. TEVES, T. TOL)

The X-ray image intensifier is an electronic tube, producing an image that is approximately 1000 times brighter than that of an ordinary fluoroscopic screen, when irradiated with the same X-ray dosage. The tube functions on the principle of the image converter: X-rays produce on a fluorescent screen an image, whose radiation liberates electrons from a photocathode in contact with the screen. The electrons focused by an electrostatic electron lens form a bright image on a second screen.

With this tube, applied in X-ray diagnostics, the radiologist can see more details than with conventional fluoroscopy, even without a long visual adaptation period.

TEVES, M. C. and TOL, T., Electronic intensification of fluoroscopic images. Philips tech. Rev. **14** [1952], 33.

Physiological department (S. WOLDRING)

A demonstration was given of testing vibration-thresholds of the skin for frequencies from 20 to 1000 c/s.

The amplitude of the mechanical vibrator is measured by integrating the inductive EMF of a separate coil connected to the vibrating stylus.

Galvanic skin responses have been used as an aid to audiometry. A psychogalvanometer built for this purpose was shown.

Respiratory physiology asks for rapid gas analysis. Several methods have been tested in this laboratory. An apparatus by which the intensity of the spectral lines of atmospheric gases in a discharge tube is measured, was shown.

Electron Microscope (A. C. VAN DORSTEN)

A new electron microscope was demonstrated. The instrument is of simplified design but still capable of resolutions better than 100 Å (10^{-6} centimetres). By reducing the focal length of the objective lens down to practically its physical limit, namely, 0.8 millimetre, a considerable reduction of the chromatic error has been obtained and it has been possible to omit the complicated electronic stabilizers for lens current and high voltage that have been common to all magnetic electron microscopes so far.

VAN DORSTEN, A. C., NIEUWDORP, H. and VERHOEFF, A., The Philips 100 kV electron microscope. Philips tech. Rev. **12** [1950], 33.

Large-screen television system (J. HAANTJES)

Part of the demonstrations were shown by means of television. The lecturers were in a small studio which was used for three years for the first experimental television transmissions in the Netherlands.

Two cameras, using image iconoscopes, took pictures of the lecturers and the objects to be shown.

The pictures were reproduced by means of a large screen projector. The projector contains a 12.5 cm projector tube working with 50 kV on its final anode and is provided with a so-called SCHMIDT optical system.

HAANTJES, J. and VAN LOON, C. J., A large-screen television projector. Philips tech. Rev. **15** [1953], 27.

FRANCKEN, J. C. and BRUINING, H., New developments in the image iconoscope. Philips tech. Rev. **14** [1953], 327.

Transistors (F. H. STIELTJES)

Transistors are new electronic amplifiers. There is no vacuum in them, nor a heated cathode. In one

type, the so-called junction-type, all actions take place in one tiny single crystal of a well-chosen semiconductor, preferably germanium at the present time. In spite of its smallness, the single crystal should contain three different layers namely one "N-layer", called the emitter, which has a function analogous to the cathode of a vacuum triode, one "P-layer" called the base, having a function analogous to that of the vacuum and the grid combined, and another "N-layer", called the collector, which acts as a sort of anode.

A circuit model incorporating a junction transistor was shown over the television system.

Delta modulation (F. DE JAGER)

In delta modulation the information signal as speech or music is converted into a sequence of binary pulses. These pulses are transmitted as 1,0 pulses which occur at equidistant time-intervals. As each pulse is known to have either the value 1 or 0, the noise in the transmission path can be eliminated if it does not exceed half the amplitude of a pulse. At the receiving end, a reconstruction of the original is obtained by applying this sequence of pulses to a linear network which consists of an integrating device and a low-pass filter. The required pulse series is obtained at the sending end by making use of a quantized feedback: a feedback action which is quantized both in time and in amplitude. Due to the process of quantization a small difference remains between the original and the reconstructed signal and this is heard as a quantizing noise. This noise can be decreased by increasing the pulse frequency.

SCHOUTEN, J. F., DE JAGER, F. and GREEFKES, J. A., Delta modulation, a new modulation system for telecommunication. Philips tech. Rev. 13 [1952], 237.

DE JAGER, F., Deltamodulation, a method of PCM transmission using the 1-unit code. Proc. London Symp. on Commun. Theory, 1952 (Butterworths scient. Publ.). Philips Res. Rep. 7 [1952], 442.

Music reproduction (D. KLEIS)

In order to trace the elements essential for good sound reproduction, eight-channel sound reproduction equipment has been installed in the Laboratories' experimental auditorium.

By means of an endless magnetic tape, six reproducing heads and feedback from the last head, reverberation is artificially produced and added to a two-channel stereophonic reproduction by means of separate loudspeakers distributed over the auditorium. This makes it possible to control and vary independently different elements that influence the sound quality.

During the demonstration the ratio of direct to indirect sound was varied, giving the illusion that the soloist was moving forward and backward. Changing the feedback results in a variation of the reverberation time. This can also be obtained by changing the speed of the tape, but the acoustic illusion is different.

The final chorus of BACH's "Mattheus Passion" directed by HENRI ARENDS was reproduced in the darkened hall with the circuit constants adjusted to simulate church acoustics. The following references relate to stereophonic reproduction only.

DE BOER, K., Stereophonic sound reproduction. Philips tech. Rev. 5 [1940], 107.

DE BOER, K. and VAN URK, A. TH., Some particulars of directional hearing. Philips tech. Rev. 6 [1941], 359.

DE BOER, K., Formation of stereophonic images. Philips tech. Rev. 8 [1946], 51.

DE BOER, K., A remarkable phenomenon with stereophonic sound reproduction. Philips tech. Rev. 9 [1947], 8.

Ferromagnetic materials (H. P. J. WIJN)

Ferroxdure is the name for a type of permanent magnetic material developed during the last few years in the Philips Research Laboratories. It is a ceramic product consisting mainly of very small hexagonal crystals of $BaFe_{12}O_{19}$. For the crystal-oriented kind of Ferroxdure the quality factor $(BH)_{max}$ can be about 3×10^6 gauss oersted.

Properties favourable for some applications were discussed.

Something was also said about the influence of mechanical stresses on the hysteresis loop of the soft magnetic ferrites; glass covered ferrite rings showing a rectangular hysteresis loop were shown.

WENT, J. J., RATHENAU, G. W., GORTER, E. W. and VAN OOSTERHOUT, G. W., Ferroxdure, a class of new permanent magnet materials. Philips tech. Rev. 13 [1952], 194.

RATHENAU, G. W., SMIT, J. and STUYTS, A. L., Z. Phys. 133 [1952], 250.

WIJN, H. P. J., Thesis Leyden 1953, Philips Laboratories separate No. 2092.

Electronic telephone exchange (J. DOMBURG)

The idea of using electronic circuits in an automatic telephone exchange is not new. Light rays, electron rays, rectifier cells and vacuum tubes, amongst other things, have been suggested for application as speech contacts.

To study the possibilities of electronic switching in automatic exchanges, a small all-electronic exchange for 10 lines and 3 connecting circuits has been designed, in which cold-cathode tubes have been used for switching purposes in both speech circuits and the signalling circuits.

Some basic principles were explained and a demonstration was given.

Six, W., Een geheel electronische automatische telefooncentrale. De Ingenieur **63** [1951], E 50.

Six, W., Het gebruik van koude kathodebuizen als schakelelement. Tijdschrift van het Nederlands Radiogenootschap **17** [1952], 17.

Six, W., Communication News (not yet published).

Domburg, J. and Six, W., Philips tech. Rev. (not yet published).

Hearing Aids (A. G. Th. Becking)

One of the hearing aid problems that is still unsolved is what its overall acoustic frequency response should be. This applies to its general behaviour, but especially to the question as to what extent peaks and valleys can be tolerated. A sort of master hearing aid has been constructed, which allows us to test, on hard-of-hearing persons, characteristics which range from a very peaked to a completely smooth response. Some of its possibilities were demonstrated over a loudspeaker.

Blom, P., An electronic hearing aid. Philips tech. Rev. **15** [1953], 37.



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